



**Avaya Solution & Interoperability Test Lab**

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**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

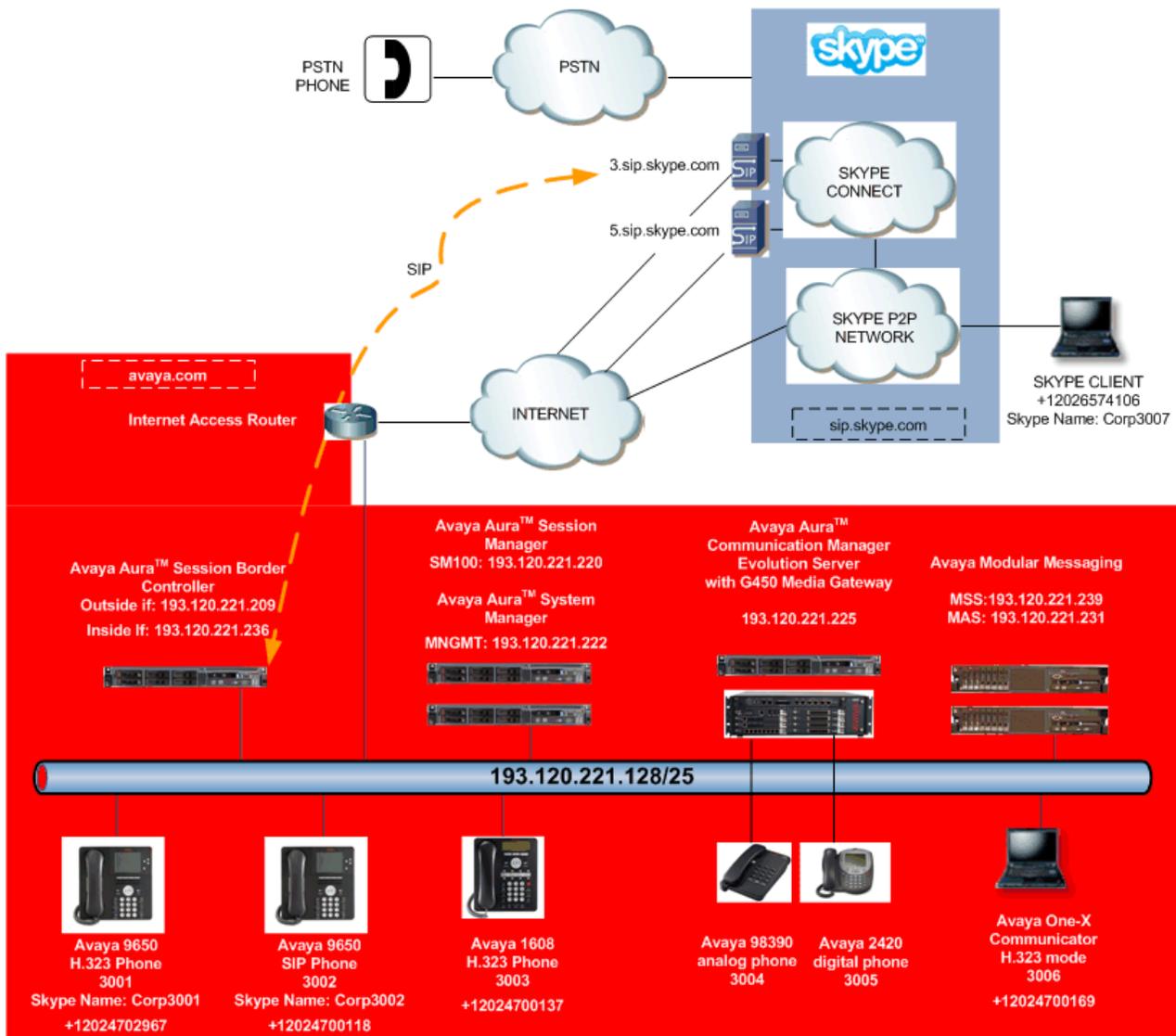
## 1.4. Known Limitations

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**

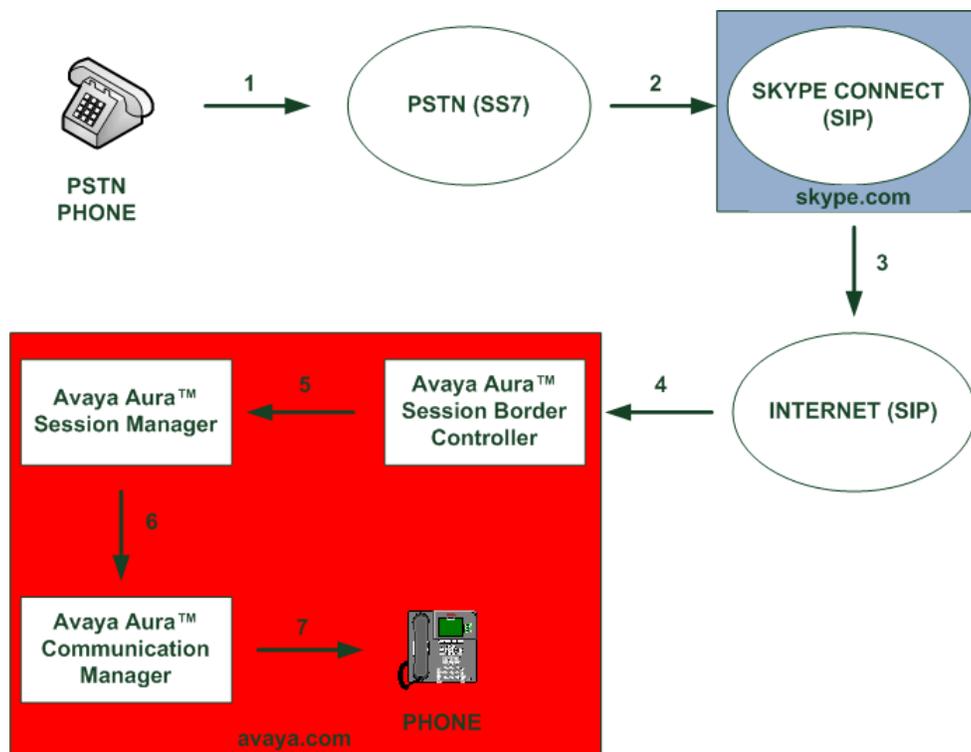
### 3. Call Flows

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.



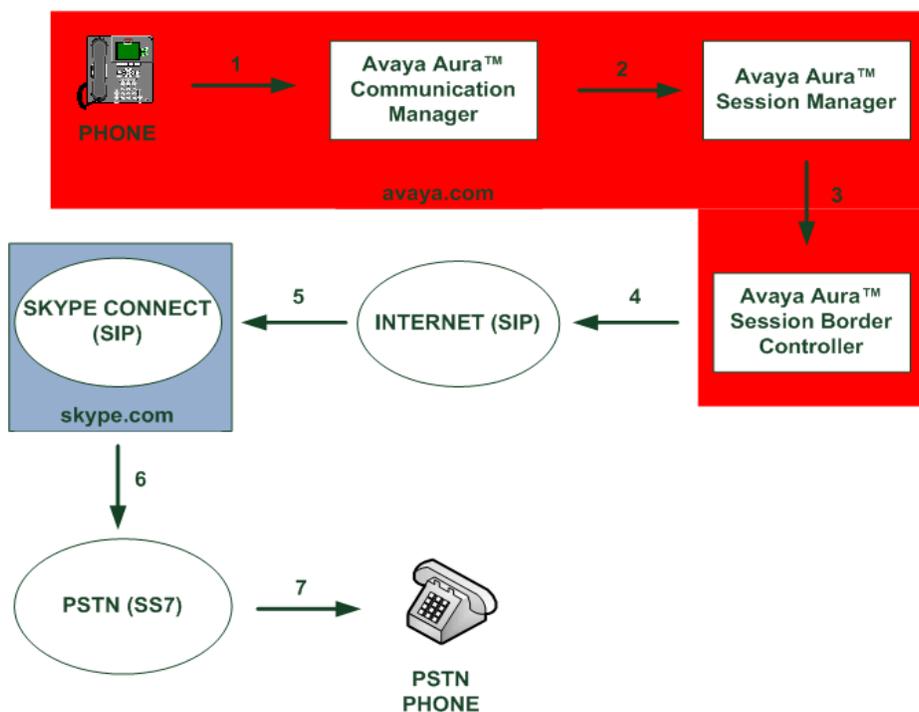
**Figure 2 - Inbound PSTN to Skype Connect Call**

**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.



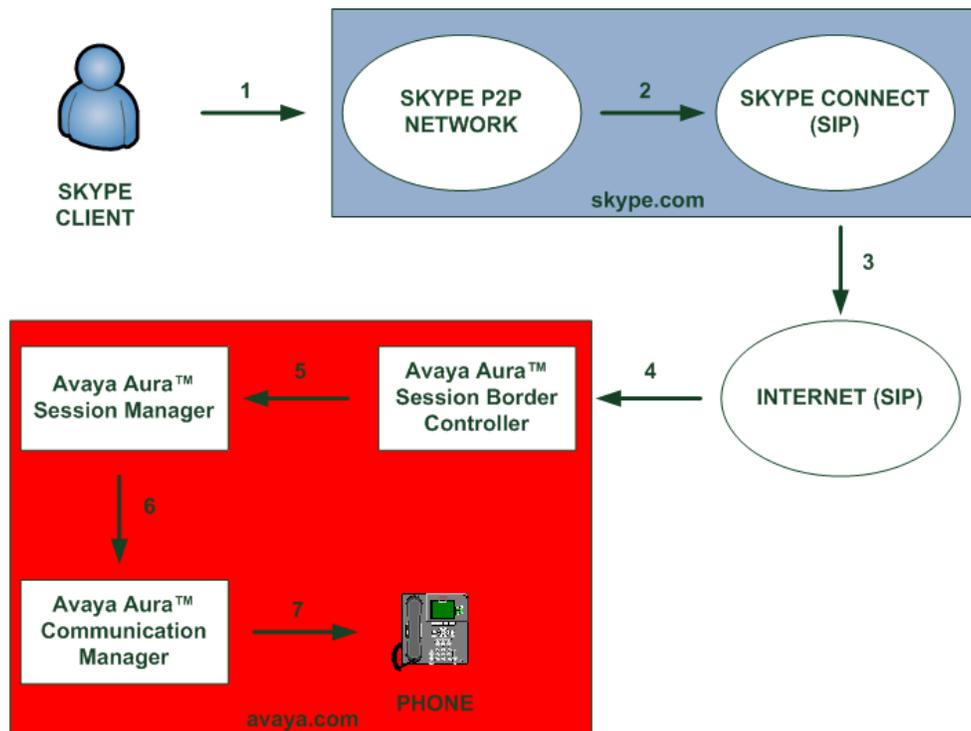
**Figure 3: Outbound Skype Connect to PSTN call with CLIP**

**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.

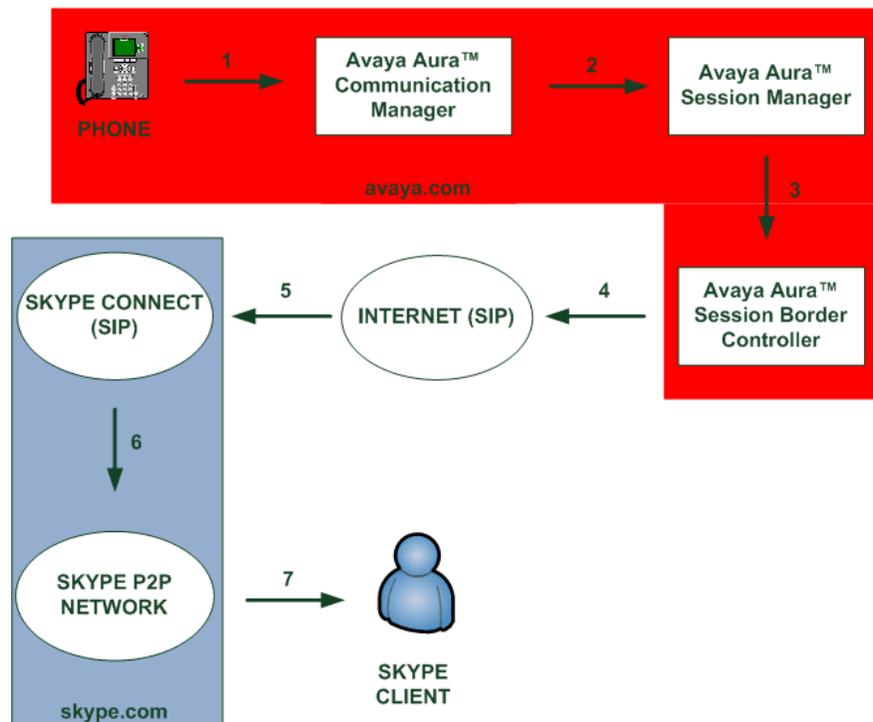


**Figure 4: Inbound Skype P2P to Skype Connect call**

### 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.



Avaya Aura™ System Manager 6.0

Home / Log On

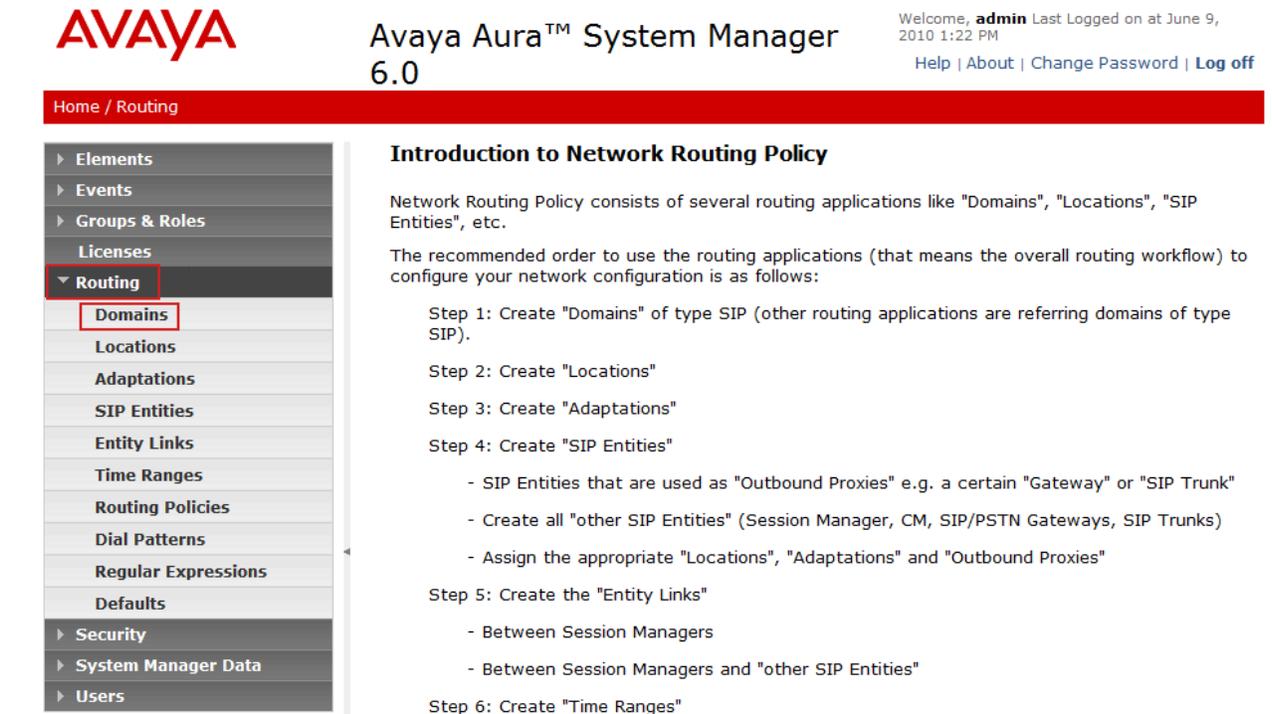
**Log On**

Username :

Password :

## 4.2. Administer SIP Domain

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



The screenshot shows the Avaya Aura System Manager 6.0 interface. The top navigation bar is red and contains the Avaya logo, the product name "Avaya Aura™ System Manager 6.0", and a user status message: "Welcome, admin Last Logged on at June 9, 2010 1:22 PM" with links for "Help | About | Change Password | Log off". Below the navigation bar is a breadcrumb trail: "Home / Routing". On the left is a sidebar menu with categories: Elements, Events, Groups & Roles, Licenses, Routing (expanded), Security, System Manager Data, and Users. Under the "Routing" category, "Domains" is highlighted with a red box. The main content area is titled "Introduction to Network Routing Policy" and contains the following text:

**Introduction to Network Routing Policy**

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"
  - SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
  - Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
  - Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
- Step 5: Create the "Entity Links"
  - Between Session Managers
  - Between Session Managers and "other SIP Entities"
- Step 6: Create "Time Ranges"

Click **New**.



The screenshot shows the Avaya Aura System Manager 6.0 interface at the "Domain Management" page. The top navigation bar is red and contains the Avaya logo, the product name "Avaya Aura™ System Manager 6.0", and a user status message: "Welcome, admin Last Logged on at June 9, 2010 1:22 PM" with links for "Help | About | Change Password | Log off". Below the navigation bar is a breadcrumb trail: "Home / Routing / Domains". On the left is a sidebar menu with categories: Elements, Events, Groups & Roles, Licenses, Routing (expanded), Security, System Manager Data, and Users. The main content area is titled "Domain Management" and contains the following text:

**Domain Management**

Edit **New** Duplicate Delete More Actions ▾

1 Item | Refresh Filter: Enable

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

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 9, 2010 1:22 PM  
Help | About | Change Password | Log off

Home / Routing / Domains

Domain Management Commit Cancel

1 Item | Refresh Filter: Enable

Name	Type	Default	Notes
* avaya.com	sip	<input type="checkbox"/>	

\* Input Required Commit Cancel

### 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**.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 9, 2010 1:22 PM  
Help | About | Change Password | Log off

Home / Routing / Locations

Location Edit New Duplicate Delete More Actions Commit

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>		

Select : All, None

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.

Home / Routing / Locations / Location Details

Location Details Commit Cancel

**General**

\* Name:  Notes:

Managed Bandwidth:  Kbit/sec

\* Average Bandwidth per Call:  Kbit/sec

**Location Pattern**

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* <input type="text" value="193.120.221.*"/>	<input type="text"/>

Select : All, None

\* Input Required Commit Cancel

Help  
Help for Locations Details fields

## 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.



Home / Routing / SIP Entities

**SIP Entities**

4 Items [Refresh](#) Filter: Enable

<input type="checkbox"/>	Name	Entity Links	FQDN or IP Address	Type	Notes
<input type="checkbox"/>					
<input type="checkbox"/>					
<input type="checkbox"/>					
<input type="checkbox"/>					

Select : All, None

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**.

**AVAYA** Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 9, 2010 1:22 PM  
[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / SIP Entities / SIP Entity Details

**SIP Entity Details** Commit Cancel

**General**

\* **Name:**

\* **FQDN or IP Address:**

**Type:**

**Notes:**

**Adaptation:**

**Location:**

**Time Zone:**

**Override Port & Transport with DNS SRV:**

\* **SIP Timer B/F (in seconds):**

**Credential name:**

**Call Detail Recording:**

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**.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 9, 2010 1:22 PM  
Help | About | Change Password | Log off

Home / Routing / SIP Entities / SIP Entity Details

**SIP Entity Details** Commit Cancel

**General**

\* Name: aasbc

\* FQDN or IP Address: 193.120.221.236

Type: SIP Trunk

Notes: AASBC - inside if

Adaptation: [v]

Location: enterprise

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV:

\* SIP Timer B/F (in seconds): 4

Credential name: [ ]

Call Detail Recording: egress

## 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**.



[Home](#) / [Routing](#) / [Entity Links](#)

▸ Elements
▸ Events
▸ Groups & Roles
Licenses
▾ Routing
Domains
Locations
Adaptations
SIP Entities
<b>Entity Links</b>
Time Ranges

### Entity Links

1 Item | [Refresh](#)

Filter: [Enable](#)

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
<input type="checkbox"/>								

Select : All, None

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**.

AVAYA Avaya Aura™ System Manager 6.0

Welcome, **admin** Last Logged on at June 9, 2010 3:32 PM  
Help | About | Change Password | Log off

Home / Routing / Entity Links

Entity Links Commit Cancel

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* toCMES	* asm	TCP	* 5060	* cmes	* 5060	<input checked="" type="checkbox"/>	full-call model non-I

\* Input Required Commit Cancel

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**.



▶ Elements
▶ Events
▶ Groups & Roles
Licenses
▼ Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links

Entity Links

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
toAASBC	* asm	TCP	* 5060	* aasbc	* 5060	<input checked="" type="checkbox"/>	

\* Input Required

## 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**.



Home / Routing / Routing Policies

▶ Elements
▶ Events
▶ Groups & Roles
Licenses
▼ Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
<b>Routing Policies</b>
Dial Patterns
Regular Expressions
Defaults

### Routing Policies

Items		Refresh	
<input type="checkbox"/>	Name	Disabled	Destinati
<input type="checkbox"/>		<input type="checkbox"/>	
<input type="checkbox"/>		<input type="checkbox"/>	
<input type="checkbox"/>		<input type="checkbox"/>	

Select : All, None

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**.

Home / Routing / Routing Policies / Routing Policy Details

**Routing Policy Details** Commit Cancel

**General**

\* Name:

Disabled:

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
cmes	193.120.221.225	CM	CM - Evolution Server R6.0

**Time of Day**

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

	Ranking <sup>1</sup>	Name <sup>2</sup>	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

Help  
Help for Routing Policy Details

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™ 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**.

Home / Routing / Routing Policies / Routing Policy Details

- ▶ Elements
- ▶ Events
- ▶ Groups & Roles
- Licenses
- ▼ Routing
  - Domains
  - Locations
  - Adaptations
  - SIP Entities
  - Entity Links
  - Time Ranges
  - Routing Policies
  - Dial Patterns
  - Regular Expressions
  - Defaults
- ▶ Security
- ▶ System Manager Data
- ▶ Users

### Routing Policy Details

**General**

\* Name:

Disabled:

Notes:

**SIP Entity as Destination**

Name	FQDN or IP Address	Type	Notes
aasbc	193.120.221.236	SIP Trunk	AASBC - inside if

**Time of Day**

1 Item Refresh											Filter: Enable	
<input type="checkbox"/>	Ranking 1 ▲	Name 2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

**Help**

[Help for Routing Policy Details fields](#)

## 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**.



▶ Elements
▶ Events
▶ Groups & Roles
Licenses
▼ Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

### Dial Patterns

4 Items [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Pattern	Min	Max	Emergency Call	SIP Domain	Notes
<input type="checkbox"/>						
<input type="checkbox"/>						
<input type="checkbox"/>						
<input type="checkbox"/>						

Select : All, None

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.

**Dial Pattern Details** Commit Cancel

**General**

\* **Pattern:** 30xx

\* **Min:** 4

\* **Max:** 4

**Emergency Call:**

**SIP Domain:** avaya.com

**Notes:**

**Originating Locations and Routing Policies**

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name <sup>1</sup>	Originating Location Notes	Routing Policy Name	Rank <sup>2</sup>	Routing Policy Disabled	Routing Policy Destination	Rou Poli Not
<input type="checkbox"/>	enterprise		toCMES-30xx	0	<input type="checkbox"/>	cmes	

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.

**Dial Pattern Details** Commit Cancel

**General**

\* **Pattern:** 1

\* **Min:** 11

\* **Max:** 11

Emergency Call:

**SIP Domain:** avaya.com

Notes:

**Originating Locations and Routing Policies**

Add Remove

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination
<input type="checkbox"/>	enterprise		toCMES-30xx	0	<input type="checkbox"/>	cmes

Select : All, None

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.

**Dial Pattern Details** Commit Cancel

**General**

\* **Pattern:** 00

\* **Min:** 2

\* **Max:** 36

**Emergency Call:**

**SIP Domain:** avaya.com

**Notes:**

**Originating Locations and Routing Policies**

Add Remove

Item	Refresh	Filter: Enable					
<input type="checkbox"/>	Originating Location Name <sup>1</sup>	Originating Location Notes	Routing Policy Name	Rank <sup>2</sup>	Routing Policy Disabled	Routing Policy Destination	Routing Policy Not
<input type="checkbox"/>	enterprise		toSkype-00	0	<input type="checkbox"/>	aasbc	

Select : All, None

## 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	
<b>Maximum Administered SIP Trunks:</b>	<b>5000</b>	<b>250</b>	
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 Codec	Silence Suppression	Frames Per Pkt	Packet 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	
<b>FAX</b>	<b>off</b>	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 formenter **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
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:

Incoming Dialog Loopbacks: eliminate      Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                 RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3        Direct IP-IP Audio Connections? y
Enable Layer 3 Test? Y                   IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n    Direct IP-IP Early 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
                                     UII 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      Total
  Len  Code     Grp(s)  Prefix  Len
  5    3001      100     12024703313  11
                                           Total Administered: 2
                                           Maximum Entries: 9999
```

## 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/ Feature	Number Len	Number Digits	Del	Insert		
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 screenshot shows the Skype Manager dashboard. At the top, the Skype Manager logo is on the left, and navigation links for Avaya, Account details, tony.skype11, Sign out, Help, and Chat support are on the right. Below the logo is a dark toolbar with icons for Dashboard, Features (highlighted with a red box), and a report icon. The current balance is displayed as €226,94, with a 'Buy Skype Credit' button and a search bar for members. The main content area is divided into 'Reports' and 'Your account'. The 'Reports' section includes an 'Allocations' bar chart showing usage from June 2009 to June 2010. The 'Your account' section shows the current balance of €226,94, a note that auto-recharge is disabled, and upcoming allocations of €206,35 for the next 30 days. Below these sections are three main cards: 'Your features' (6 members have Skype Credit), 'Your Members' (13 members, with an 'Add Members' button), and 'News' (a welcome message about the new product).

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Dashboard Features €226,94 Buy Skype Credit Search Members

Reports Your account Account status: All OK ✓

**Allocations**

Month	Allocations
Jun 2009	0
Jul 2009	0
Aug 2009	0
Sep 2009	0
Oct 2009	0
Nov 2009	0
Dec 2009	0
Jan 2010	0
Feb 2010	0
Mar 2010	0
Apr 2010	~50
May 2010	~350
Jun 2010	~200

**Current balance** €226,94 Auto-recharge is disabled

**Upcoming allocations** €206,35 (next 30 days) Review allocations

Your account will be automatically recharged when payments are due.

**Your features** 6 members have Skype Credit

**Your Members** Your Skype Manager has 13 members Add Members

**News** Welcome! Skype Manager is a brand new product that replaces the Business Control Panel. From setting up employee accounts, to allocating Skype

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

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€226,94 Buy Skype Credit Search Members

**Skype for SIP** 3 profiles

### Credit allocations

Skype Credit allows members to call phones, send SMS, use a Skype To Go number and other features. [Learn more](#)

Filter this list

<input type="checkbox"/>	Name ^	Credit	Auto-recharge
<input type="checkbox"/>	Avaya.SILWestminster2 Avaya.SILWestmins	€0,00	Auto-recharge disabled
<input type="checkbox"/>	Avaya.SILWestminster Avaya.SILWestminst	€0,00	Auto-recharge disabled
<input type="checkbox"/>	corp3001	€0,00	Auto-recharge disabled
<input type="checkbox"/>	corp3002	€0,00	Auto-recharge disabled
<input type="checkbox"/>	corp3007	€0,00	Auto-recharge disabled
<input type="checkbox"/>	dublinsil1	\$6.56	Auto-recharge disabled
<input type="checkbox"/>	dublinsil2	€9,94	Auto-recharge active

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

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€226,94 Buy Skype Credit Search Members

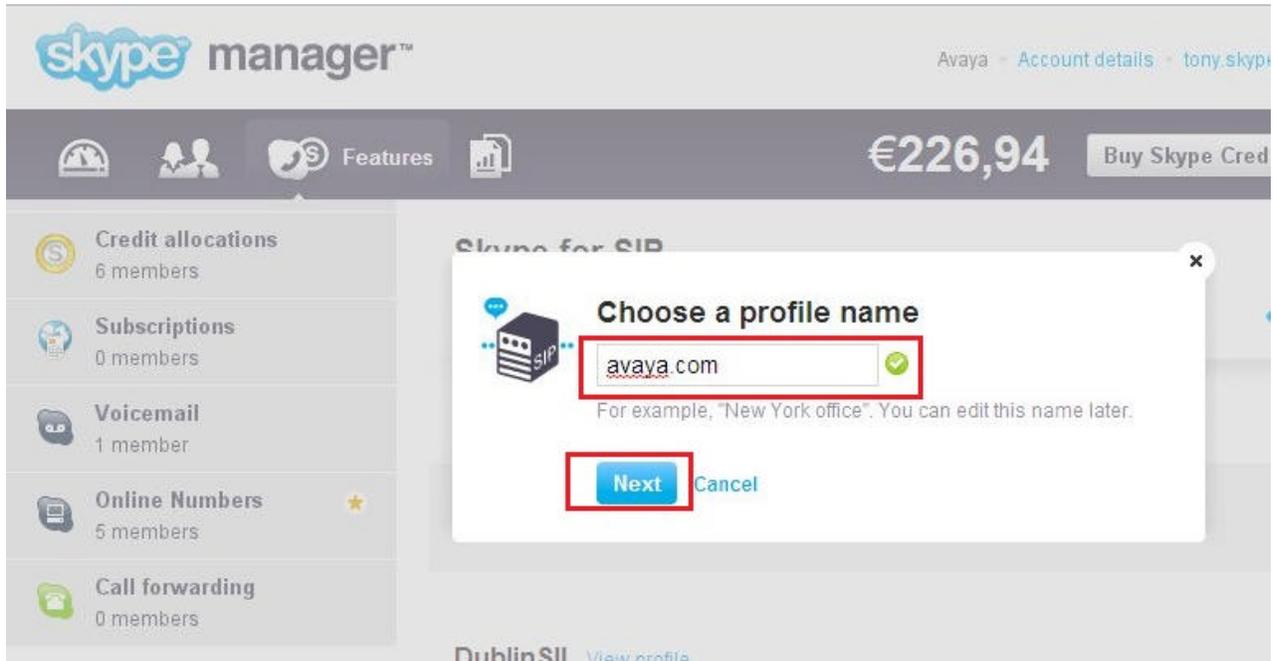
### Skype for SIP

Connect your existing SIP-enabled PBX to Skype with Skype for SIP. [Learn more](#)

#### Your SIP Profiles

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.

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Dashboard Features €226,94 Buy Skype Credit Search Members

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

#### Authentication details

Please choose the method of authentication needed for your PBX.

Registration (Username/password)  or, IP Authentication

SIP User	99051000109702
Password	VCC8yjAQqBvswC <a href="#">Generate a new password</a>
Skype for SIP address	sip.skype.com
UDP Port	5060

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

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€226,94 Buy Skype Credit

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

### Authentication details

Please choose the method of authentication needed for your PBX.

Registration (Username/password) or,  IP Authentication

Please enter the IP details for your PBX

Public IP address	193.120.221.209
UDP Port	5060

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**.

The screenshot displays the configuration interface for a SIP profile. On the left is a navigation sidebar with 'avaya.com' selected, and options for 'Profile settings', 'Authentication details', and 'Reports'. The main content area is titled 'Please choose the method of authentication needed for your PBX.' and offers two options: 'Registration (Username/password)' and 'or, IP Authentication'. Below this, a section titled 'Your PBX details' lists: SIP User: 99051000109702, Public IP address: 193.120.221.209, and UDP Port: 5060. A 'Change PBX details' link is provided. The final section, 'Use these details to configure your PBX', lists 'Skype for SIP addresses' with Primary: 3.sip.skype.com and Secondary: 5.sip.skype.com, and 'Skype for SIP IP addresses' with Primary: 193.120.218.68 and Secondary: 78.141.179.70. A note indicates to enable traffic for these IP addresses in the firewall.

## 6.4. Administer Maximum Simultaneous Calls

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

The screenshot shows the Skype Manager interface. At the top, there's a navigation bar with the Skype Manager logo, user information (Avaya, Account details, tony.skype11, Sign out, Help, Chat support), and account balance (€226,94). Below this is a dark header with icons for home, contacts, features, and a search bar. The main content area is titled 'Profile settings' and features a sidebar on the left with navigation options: Profile settings (selected), Authentication details, Reports, and a link to 'Back to SIP Profile list'. The main settings table includes:

Setting	Value/Action
Profile name	avaya.com
Calling channels	<a href="#">Buy a channel subscription to activate this profile</a>
Outgoing calls	<a href="#">Set up outgoing calls</a>
Caller ID	<a href="#">Set up Caller ID</a>
Incoming calls	<a href="#">Add a number or business account</a>

At the bottom of the settings area, there is a button labeled 'Delete 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**.

**Channel subscription**

Calling channels cost **€4,95** / month per channel.

**Please choose the number of channels**

Number of channels required  (max. 300)  
[How many concurrent channels does my company need?](#) ▾

**Total cost**

Cost per channel	€4,95
Number of channels	3

**Total cost every month €14,85**  
The cost will be deducted from the Skype Credit balance of your Skype Manager.

By clicking 'Buy now', you agree to Skype's [Terms of Service](#).

**Buy now** [Back](#)

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

The screenshot shows the Skype Manager interface. At the top, the Skype Manager logo is on the left, and the user's account information (Avaya, Account details, tony.skype11, Sign out, Help, Chat support) is on the right. Below the header, there is a dark navigation bar with icons for home, contacts, features, and a balance of €212,09. A 'Buy Skype Credit' button and a search bar are also present. The left sidebar contains a list of features: Credit allocations (6 members), Subscriptions (0 members), Voicemail (1 member), Online Numbers (5 members), Call forwarding (0 members), and Skype for SIP (4 profiles). The main content area is titled 'Skype for SIP' and includes a sub-header 'Connect your existing SIP-enabled PBX to Skype with Skype for SIP. Learn more' and a SIP server icon. A green notification box with a checkmark states: 'Subscription changed successfully for Profile avaya.com. The profile's updated subscription setting will be available in a few minutes.' Below this, the 'Your SIP Profiles' section shows a 'Create a new profile' button and a table for the profile 'avaya.com'. The table has three rows: 'Channels' with a link 'Buy a channel subscription to activate this profile', 'Outgoing calls' with a button 'Set up outgoing calls' (highlighted with a red box), and 'Incoming calls' with a link 'Set up incoming calls'.

skype manager™ Avaya · Account details · tony.skype11 · Sign out · Help · Chat support

€212,09 Buy Skype Credit Search Members

**Skype for SIP**  
Connect your existing SIP-enabled PBX to Skype with Skype for SIP. [Learn more](#)

✓ Subscription changed successfully for Profile avaya.com  
The profile's updated subscription setting will be available in a few minutes.

**Your SIP Profiles**  
[Create a new profile](#)

**avaya.com** [View profile](#)

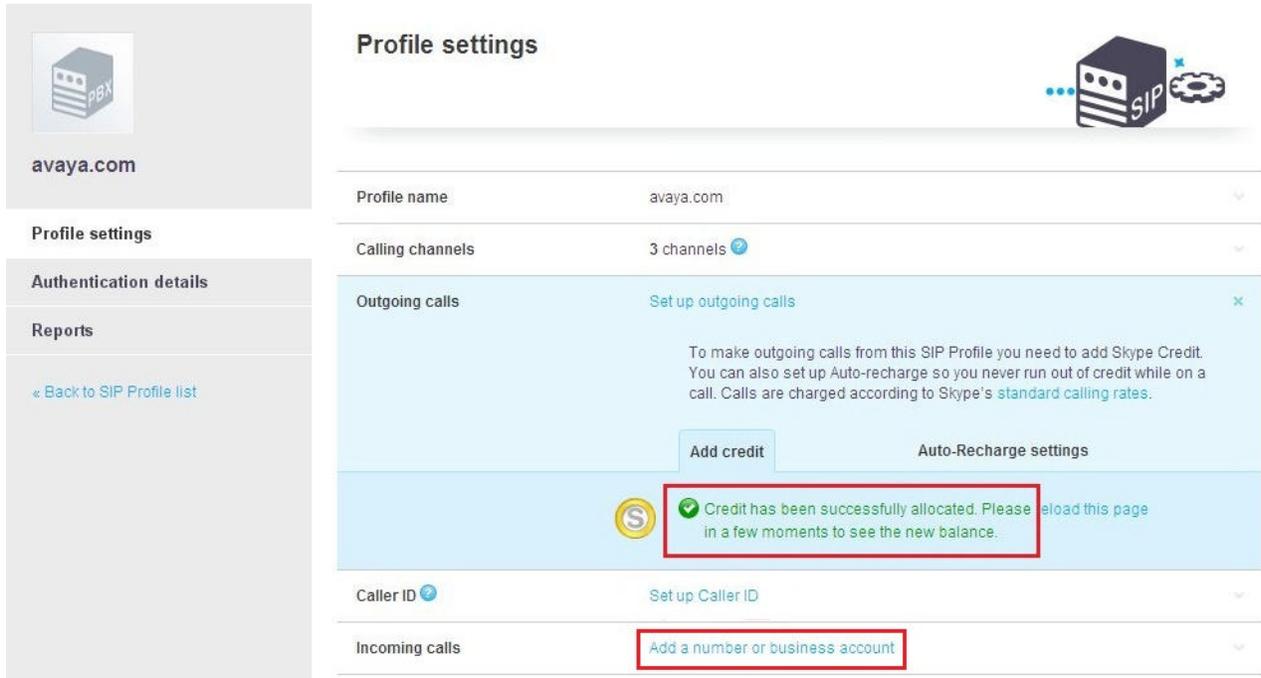
Channels	<a href="#">Buy a channel subscription to activate this profile</a>
Outgoing calls	<a href="#">Set up outgoing calls</a>
Incoming calls	<a href="#">Set up incoming calls</a>

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**.

The screenshot shows the Skype Manager interface for a user named 'avaya.com'. The top navigation bar includes the Skype Manager logo, account details, and a balance of €212,09. The main content area is titled 'Profile settings' and contains several sections: 'Profile name' (avaya.com), 'Calling channels' (3 channels), and 'Outgoing calls' (Set up outgoing calls). The 'Outgoing calls' section includes a warning message: 'To make outgoing calls from this SIP Profile you need to add Skype Credit. You can also set up Auto-recharge so you never run out of credit while on a call. Calls are charged according to Skype's standard calling rates.' Below this message are two buttons: 'Add credit' and 'Auto-Recharge settings'. At the bottom of the 'Add credit' section, there is a text input field containing '10.00' and an 'Add credit' button. The input field and the button are highlighted with red boxes.

## 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**.



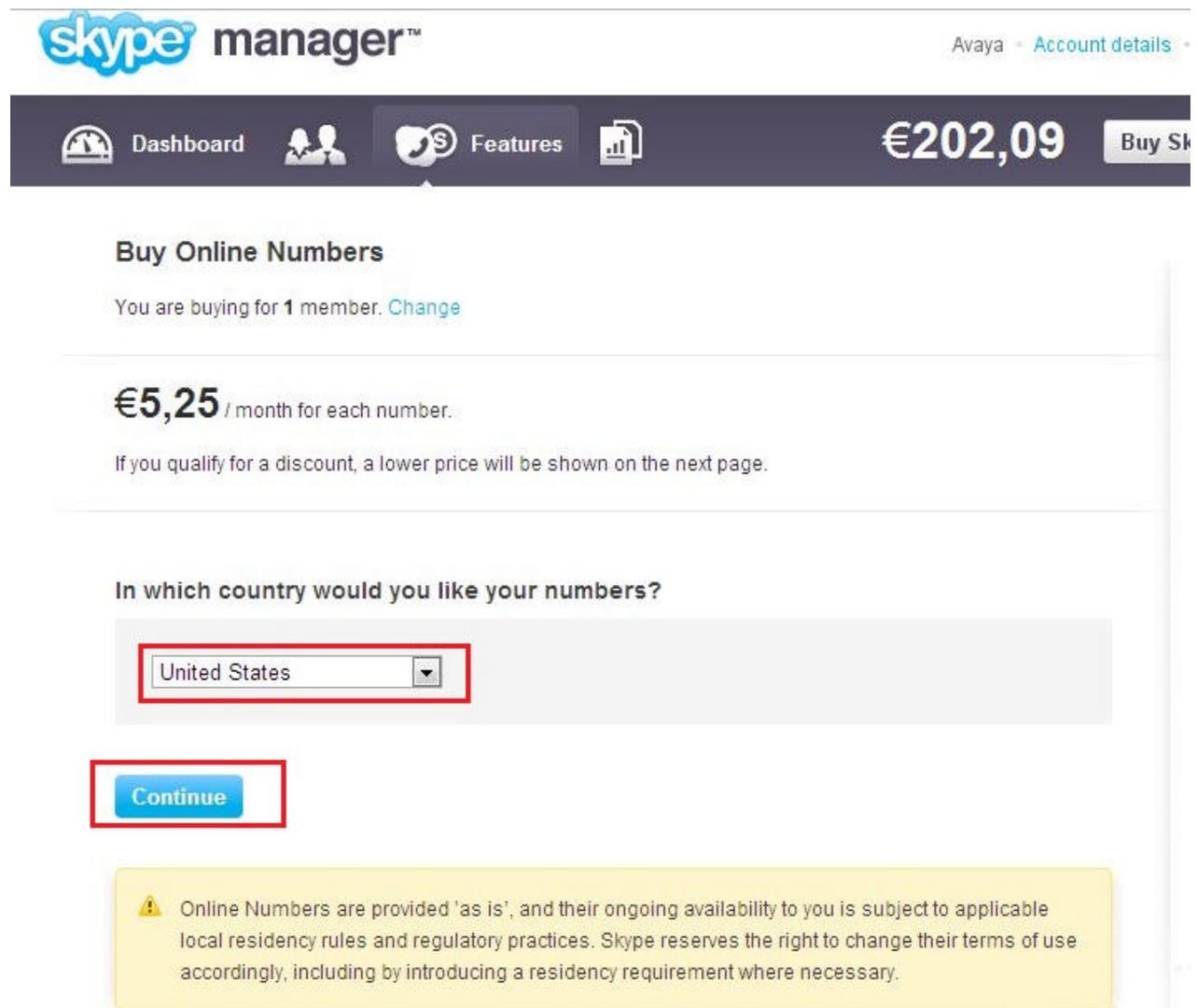
The screenshot displays the 'Profile settings' page for a SIP profile named 'avaya.com'. The left sidebar contains navigation options: 'Profile settings', 'Authentication details', and 'Reports', along with a 'Back to SIP Profile list' link. The main content area shows settings for 'Profile name', 'Calling channels', 'Outgoing calls', 'Caller ID', and 'Incoming calls'. A prominent message in the 'Outgoing calls' section states: 'Credit has been successfully allocated. Please reload this page in a few moments to see the new balance.' This message is enclosed in a red rectangular box. Below the message are buttons for 'Add credit' and 'Auto-Recharge settings'. The 'Incoming calls' section at the bottom features a button labeled 'Add a number or business account', which is also highlighted with a red rectangular box.

Profile name	avaya.com
Calling channels	3 channels
Outgoing calls	Set up outgoing calls
Caller ID	Set up Caller ID
Incoming calls	Add a number or business account

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

The screenshot shows the 'Profile settings' page for a SIP profile named 'avaya.com'. The left sidebar contains navigation options: 'avaya.com', 'Profile settings', 'Authentication details', 'Reports', and a link to 'Back to SIP Profile list'. The main content area is titled 'Profile settings' and includes a SIP icon with a gear. Below this is a list of settings: Profile name (avaya.com), Calling channels (3 channels), Outgoing calls (€10,00 Auto-recharge disabled), and Caller ID (Set up Caller ID). The 'Incoming calls' section is expanded, displaying instructions on receiving calls via Skype Online Numbers or business accounts. It features two buttons: 'Add Online Number' and 'Add business account'. Under 'Add a number', there is a dropdown menu labeled 'Select a number' and two buttons: 'Allocate number' and 'Buy a new number'. The 'Buy a new number' button is highlighted with a red rectangular border.

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



skype manager™ Avaya - Account details

Dashboard Features €202,09 Buy Sk

### Buy Online Numbers

You are buying for 1 member. [Change](#)

**€5,25** / month for each number.

If you qualify for a discount, a lower price will be shown on the next page.

In which country would you like your numbers?

United States

[Continue](#)

**!** Online Numbers are provided 'as is', and their ongoing availability to you is subject to applicable local residency rules and regulatory practices. Skype reserves the right to change their terms of use accordingly, including by introducing a residency requirement where necessary.

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).

## Buy Online Numbers

You are buying for 1 member. [Change](#)

**€5,25** / month for each number.

You have selected  United States [Change](#)

### Please choose your Online Numbers

Buy  numbers. (max: 100)

Region code

Area code

#### Click an Online Number to select it:

**+12024703313**

+12024703328

+12024703524

+12024706034

+12024706759

+12024706779

+12024706782

+12024706792

+12024706795

+12024706798

[Show 10 different Online Numbers](#)

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

The screenshot shows the Skype interface with a dark header bar. On the left is a sidebar with a profile icon and the name 'avaya.com'. Below the icon are menu items: 'Profile settings', 'Authentication details', and 'Reports'. A link '« Back to SIP Profile list' is also present. The main area is titled 'Profile settings' and features a SIP icon with a gear. Below this is a list of settings:

Profile name	avaya.com	▼
Calling channels	3 channels	🔗 ▼
Outgoing calls	€10,00 Auto-recharge disabled	▼
Caller ID	Set up Caller ID	🔗 ▼
Incoming calls	Add a number or business account	▼

At the bottom of the settings list is a button labeled 'Delete this profile'.

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

**Profile settings**

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile name	avaya.com
Calling channels	3 channels
Outgoing calls	€10,00 Auto-recharge disabled
Caller ID	Set up Caller ID

**Incoming calls** Add a number or business account

You can receive incoming calls on your SIP Profile via [Skype Online Numbers](#) and via [Skype business accounts](#). When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.

Add Online Number Add business account

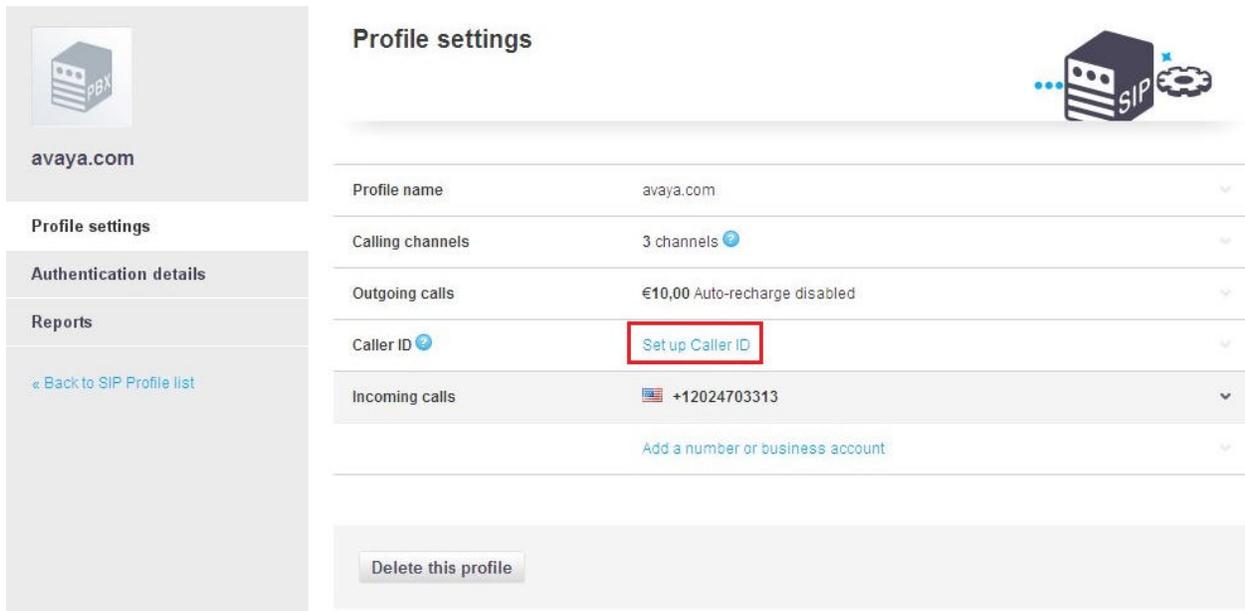
Add a number

+12024703313

Allocate number Buy a new number

## 6.6. Administer Caller ID

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



**Profile settings**

avaya.com

**Profile settings**

**Authentication details**

**Reports**

[Back to SIP Profile list](#)

Profile name	avaya.com
Calling channels	3 channels
Outgoing calls	€10,00 Auto-recharge disabled
Caller ID	<a href="#">Set up Caller ID</a>
Incoming calls	+12024703313
	<a href="#">Add a number or business account</a>

[Delete this profile](#)

Click **Use and Online Number**.



avaya.com

**Profile settings**

**Authentication details**

**Reports**

[« Back to SIP Profile list](#)

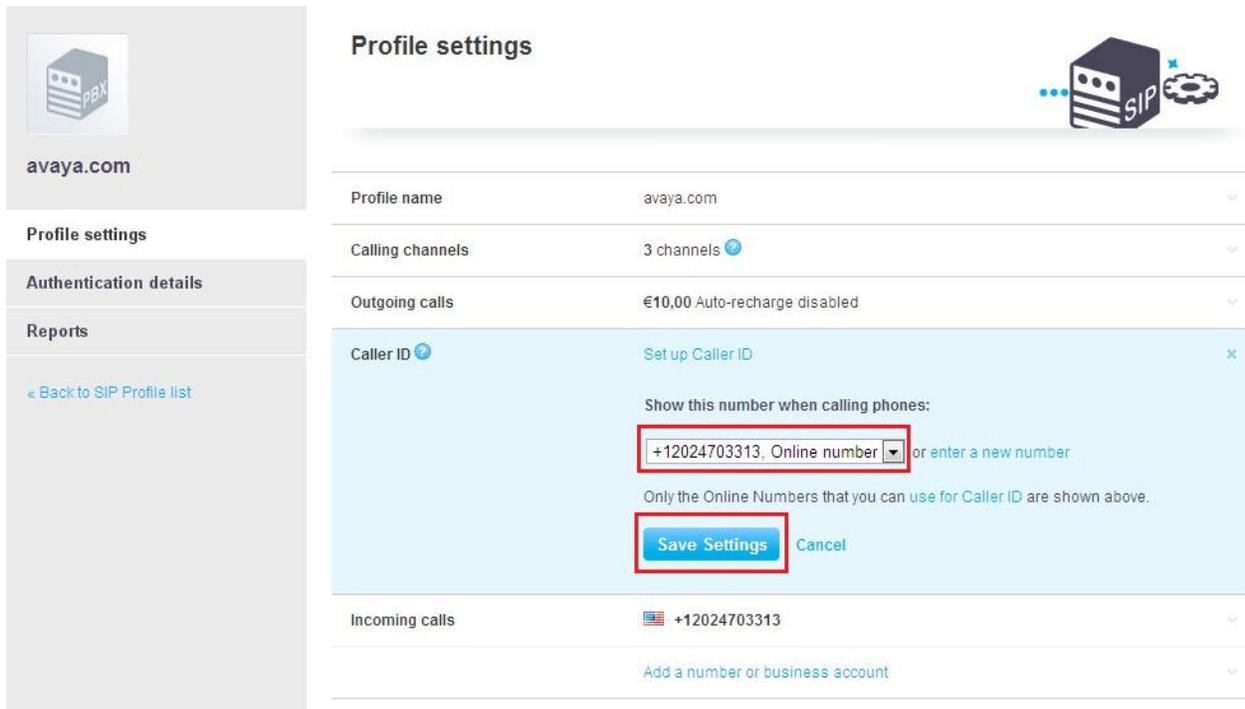
### Profile settings



Profile name	avaya.com	▼
Calling channels	3 channels	ⓘ ▼
Outgoing calls	€10,00 Auto-recharge disabled	▼
Caller ID ⓘ	Set up Caller ID <span>✕</span>	
	You can either use Online Numbers assigned to this profile or landline numbers for caller identification.	
	<a href="#">Use an Online Number</a>	<a href="#">Use a landline number</a>
Incoming calls	 +12024703313	▼
	<a href="#">Add a number or business account</a>	▼

[Delete this profile](#)

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



The screenshot displays the Avaya Profile settings interface. On the left is a sidebar with navigation options: avaya.com, Profile settings (selected), Authentication details, Reports, and a link to the SIP Profile list. The main content area is titled 'Profile settings' and features a SIP icon. A modal window for 'Caller ID' is open, showing a dropdown menu with '+12024703313, Online number' selected. Below the dropdown is a 'Save Settings' button, which is highlighted with a red box. Other settings visible include Profile name (avaya.com), Calling channels (3 channels), and Outgoing calls (€10,00 Auto-recharge disabled). The Incoming calls section shows '+12024703313' with a US flag icon and an option to 'Add a number or business account'.

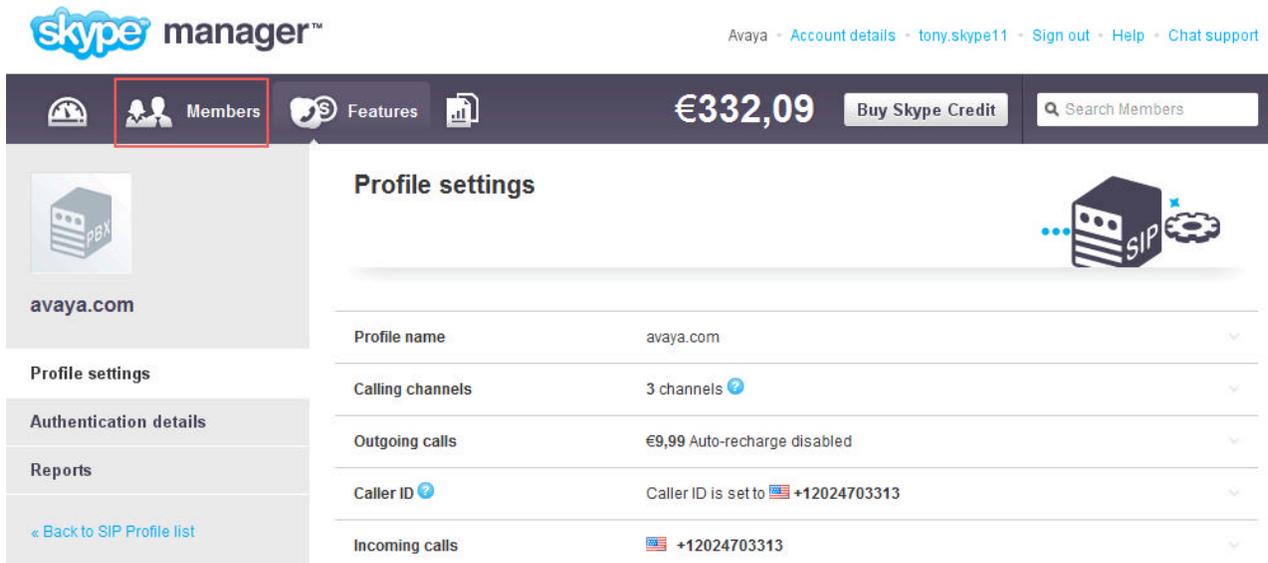
The following screenshot displays a sample Profile with both the Caller ID and the Incoming calls (DID) set to the same Online Number.

The screenshot shows the 'Profile settings' page for a profile named 'avaya.com'. The interface includes a left sidebar with navigation options: 'avaya.com', 'Profile settings', 'Authentication details', and 'Reports'. A link '« Back to SIP Profile list' is also present. The main content area is titled 'Profile settings' and features a SIP icon. A red box highlights the 'Caller ID' and 'Incoming calls' sections. The 'Caller ID' section shows 'Caller ID is set to +12024703313' with a 'Change Caller ID' button and a 'Disable Caller ID' link. The 'Incoming calls' section shows '+12024703313' with an 'Add a number or business account' link. A 'Delete this profile' button is located at the bottom of the settings area.

Setting	Value
Profile name	avaya.com
Calling channels	3 channels
Outgoing calls	€10,00 Auto-recharge disabled
Caller ID	Caller ID is set to +12024703313
Incoming calls	+12024703313

## 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.



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Members Features €332,09 Buy Skype Credit Search Members

avaya.com

### Profile settings

Profile name avaya.com

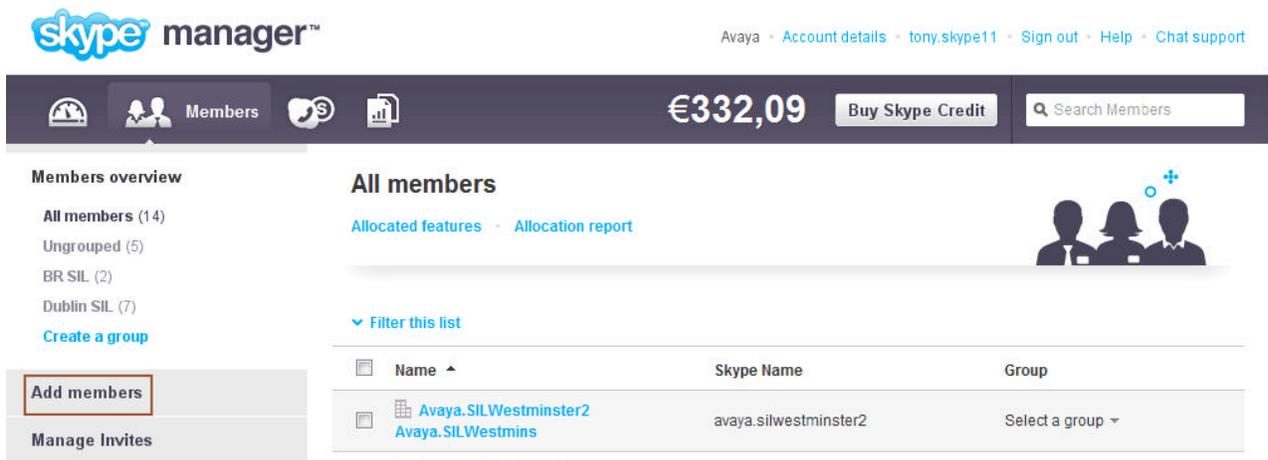
Calling channels 3 channels

Outgoing calls €9,99 Auto-recharge disabled

Caller ID Caller ID is set to +12024703313

Incoming calls +12024703313

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



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### All members

Allocated features · Allocation report

Filter this list

Name	Skype Name	Group
Avaya.SILWestminster2 Avaya.SILWestmins	avaya.silwestminster2	Select a group

Add members Manage Invites

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

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Members €332,09 Buy Skype Credit Search Members

Members overview

Add members

- Create business accounts
- Invite personal members by email
- Invite personal members by Skype Name

Manage Invites

There are two ways to add your colleagues to your Skype Manager

**Create business accounts for them**

Invite them with their personal accounts

Create business accounts Invite by email or by Skype Name

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**.

Members €332,09 Buy Skype Credit Search Members

Members overview

Add members

- Create business accounts
- Invite personal members by email
- Invite personal members by Skype Name

Manage Invites

**Create business accounts**

1 Enter email addresses/Import CSV 2 Enter details 3 Summary

Enter email addresses

bob@avaya.com

Please enter one or more email addresses of people you want to invite to join your Skype Manager.

or, import a CSV file with member data

Choose File No file chosen

The CSV file needs to contain columns for: **First name**, **Last name** and **Email address**. Optionally you can also add column for Password. [Learn how to generate the CSV file.](#)

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**.

**Members overview** ▾

**Add members**

- Create business accounts
- Invite personal members by email
- Invite personal members by Skype Name

**Manage Invites**

### Create business accounts

1 Enter email addresses 2 Enter details 3 Summary

There is some required information missing. Please fill in the fields marked with red.

We found 1 valid email addresses in your input

We've done a little magic to suggest some Skype Names based on the information you entered. Just click on a suggested name to edit it. If everything is fine, click 'Create accounts'.

Email address\*  ✓

Skype Name\*  ✓

First name

Last name

Password  ✓

The password needs to be at least 6 characters long and contain at least 1 number.

Repeat password  ✓

[Save and close](#) [Remove this account](#)

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

Members overview

Add members

- Create business accounts
- Invite personal members by email
- Invite personal members by Skype Name

Manage Invites

## Create business accounts

1 Enter email addresses 2 Enter details 3 Summary

There is some required information missing. Please fill in the fields marked with red.

We found 1 valid email addresses in your input

We've done a little magic to suggest some Skype Names based on the information you entered. Just click on a suggested name to edit it. If everything is fine, click 'Create accounts'.

Email address*	Skype Name*
bob@avaya.com	bob.avaya.com

[Add another account](#)

Add members to a group after their account is created  or [create a group](#)

**Create accounts**

## 6.8. Add Business User to SIP Profile

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

Members overview

Members

€332,09 Buy Skype Credit Search Members

### Create business accounts

1 Enter email addresses 2 Enter details 3 Summary

**The following 1 accounts were created**  
An email was sent out to the email addresses below inviting the account holders to set up their passwords.

<input type="checkbox"/>	Name	Skype Name	Email
<input checked="" type="checkbox"/>	bob.smith	bob.avaya.com	bob@avaya.com

Add to Skype contact list

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

400  
200  
0

Jun Jul Aug Sep Oct Nov Dec Jan Feb Mar Apr May Jun  
2009 2010

**Upcoming allocations**  
? (next 30 days)

### Your features

- 6 members have [Skype Credit](#)
- Set up [Subscriptions](#) for your members
- 5 members have [Online Numbers](#)
- Set up [Call forwarding](#) for your members
- 1 member has [Voicemail](#)
- 4 profiles set up for [Skype for SIP](#)

### Your Members

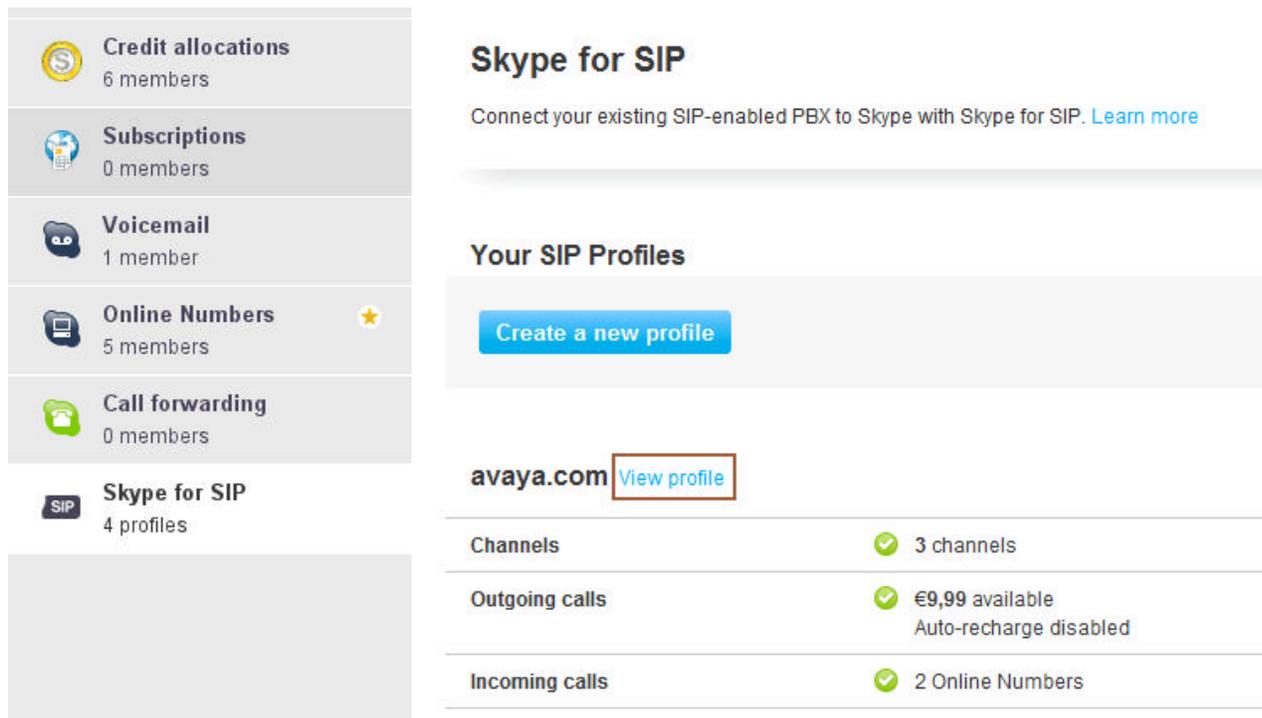
Your Skype Manager has [15 members](#)

[Add Members](#)

**Since you last signed in**  
No changes since you last logged in.

**Still outstanding**  
0 outstanding invites

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



**Credit allocations**  
6 members

**Subscriptions**  
0 members

**Voicemail**  
1 member

**Online Numbers** ★  
5 members

**Call forwarding**  
0 members

**Skype for SIP**  
4 profiles

## Skype for SIP

Connect your existing SIP-enabled PBX to Skype with Skype for SIP. [Learn more](#)

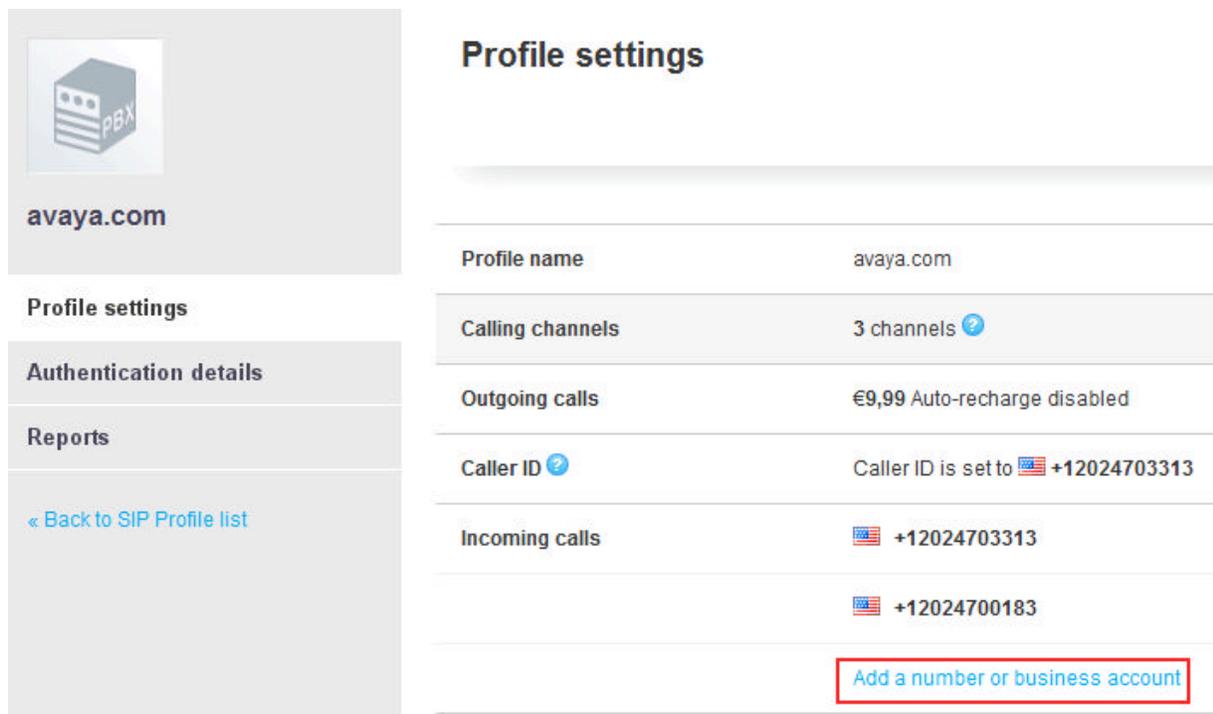
### Your SIP Profiles

[Create a new profile](#)

**avaya.com** [View profile](#)

Channels	✓ 3 channels
Outgoing calls	✓ €9,99 available Auto-recharge disabled
Incoming calls	✓ 2 Online Numbers

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



**avaya.com**

**Profile settings**

**Authentication details**

**Reports**

[« Back to SIP Profile list](#)

## Profile settings

Profile name	avaya.com
Calling channels	3 channels ?
Outgoing calls	€9,99 Auto-recharge disabled
Caller ID ?	Caller ID is set to  +12024703313
Incoming calls	+12024703313 +12024700183

[Add a number or business account](#)

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

**avaya.com**

**Profile settings**

**Authentication details**

**Reports**

[← Back to SIP Profile list](#)

Profile name	avaya.com
Calling channels	3 channels <a href="#">?</a>
Outgoing calls	€9,99 Auto-recharge disabled
Caller ID <a href="#">?</a>	Caller ID is set to  +12024703313
Incoming calls	+12024703313
	+12024700183

**Add a number or business account** ×

You can receive incoming calls on your SIP Profile via [Skype Online Numbers](#) and via [Skype business accounts](#). When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.

**Add Online Number** **Add business account**

**Add a number**

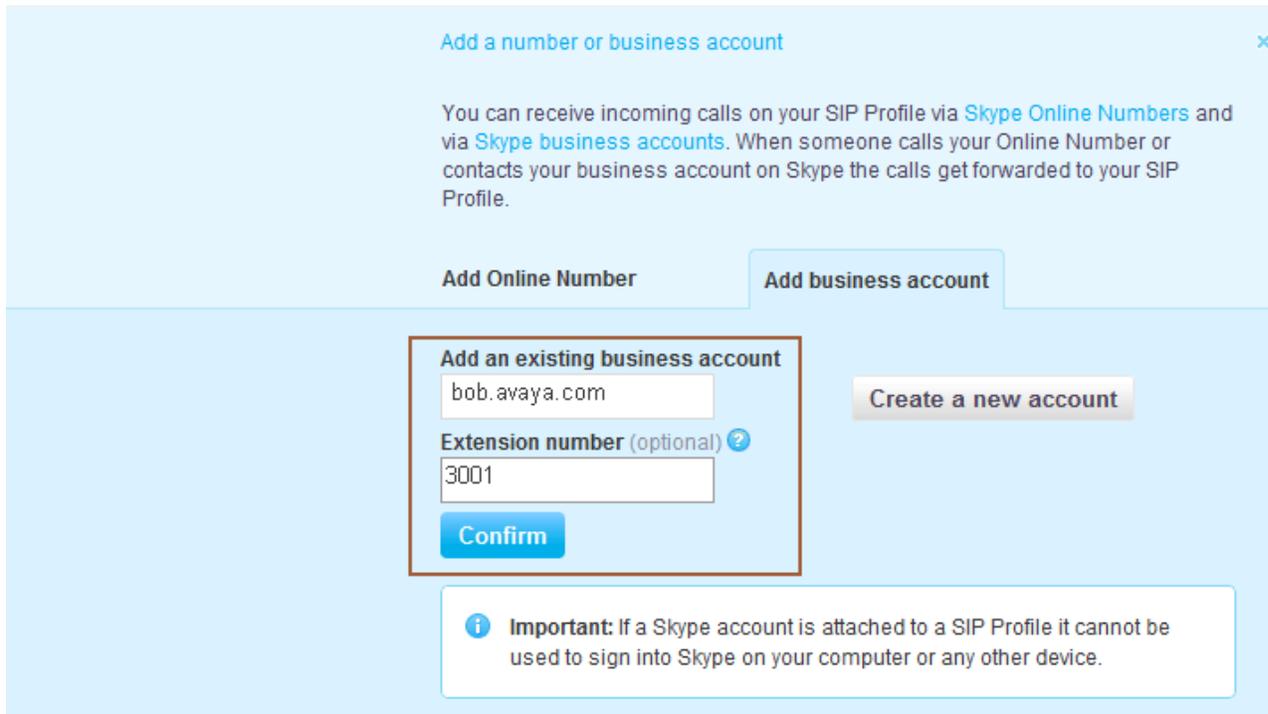
Select a number

**Allocate number** **Buy a new number**

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**.



**Add a number or business account** ×

You can receive incoming calls on your SIP Profile via [Skype Online Numbers](#) and via [Skype business accounts](#). When someone calls your Online Number or contacts your business account on Skype the calls get forwarded to your SIP Profile.

**Add Online Number** **Add business account**

**Add an existing business account**  
bob.avaya.com

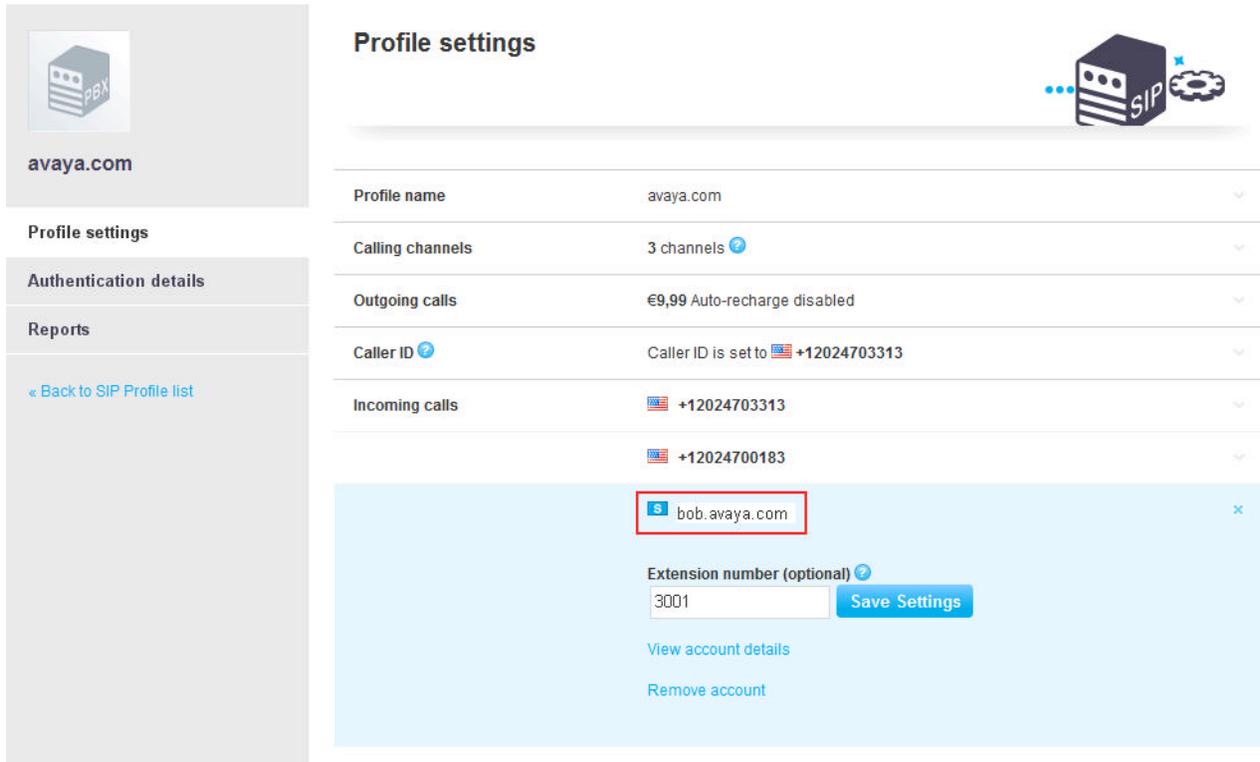
**Extension number (optional)** ?  
3001

**Confirm**

**Create a new account**

**i Important:** If a Skype account is attached to a SIP Profile it cannot be used to sign into Skype on your computer or any other device.

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



**Profile settings**

avaya.com

Profile settings

Authentication details

Reports

[« Back to SIP Profile list](#)

Profile name	avaya.com
Calling channels	3 channels
Outgoing calls	€9,99 Auto-recharge disabled
Caller ID	Caller ID is set to  +12024703313
Incoming calls	+12024703313
	+12024700183

bob.avaya.com

Extension number (optional)

3001 [Save Settings](#)

[View account details](#)

[Remove account](#)

## 7. Configure Avaya Aura™ Session Border Controller

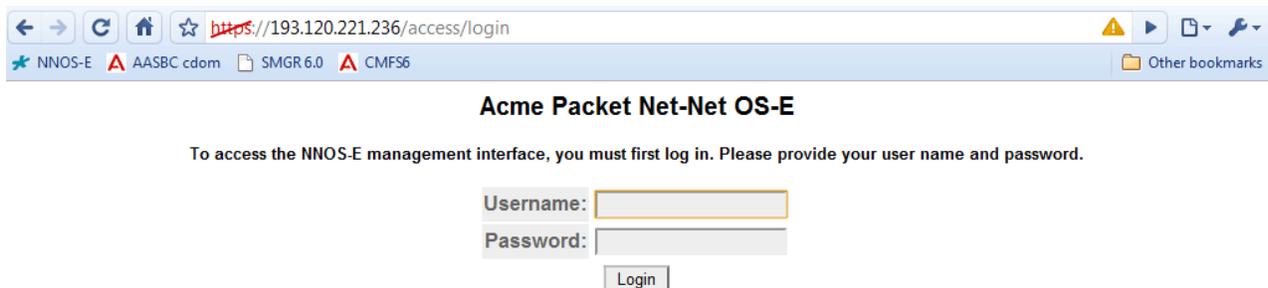
This section provides the procedures for configuring Avaya Aura™ Session Border Controller and includes the following items:

- 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.



Acme Packet Net-Net OS-E

To access the NNOS-E management interface, you must first log in. Please provide your user name and password.

Username:

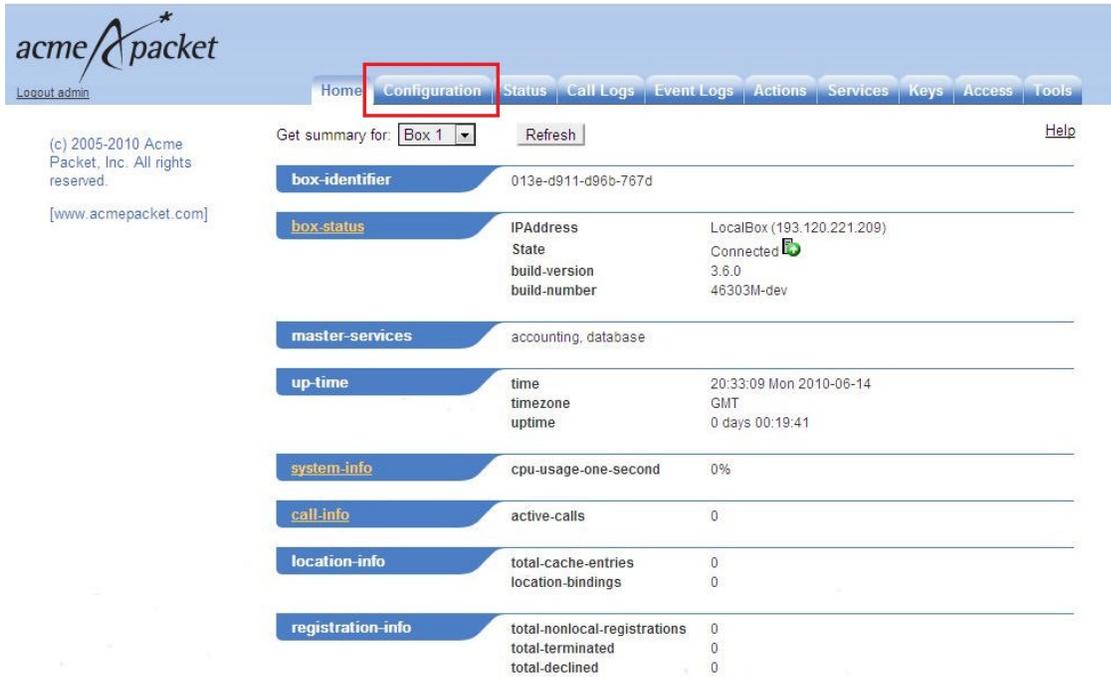
Password:

Login

## 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.



acme packet  
Logout admin

Home **Configuration** Status Call Logs Event Logs Actions Services Keys Access Tools

Get summary for:  Refresh [Help](#)

<b>box-identifier</b>	013e-d911-d96b-767d	
<b>box-status</b>	IPAddress	LocalBox (193.120.221.209)
	State	Connected 
	build-version	3.6.0
	build-number	46303M-dev
<b>master-services</b>	accounting, database	
<b>up-time</b>	time	20:33:09 Mon 2010-06-14
	timezone	GMT
	uptime	0 days 00:19:41
<b>system-info</b>	cpu-usage-one-second	0%
<b>call-info</b>	active-calls	0
<b>location-info</b>	total-cache-entries	0
	location-bindings	0
<b>registration-info</b>	total-nonlocal-registrations	0
	total-terminated	0
	total-declined	0

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[www.acmepacket.com]

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

### Configuration: all

Configuration	Setup	View
---------------	-------	------

- [-] cluster
  - + box:aasbc
- [-] vsp
  - + default-session-config
  - + pre-session-config
  - + session-config-pool
  - + dial-plan
    - registration-plan
  - [-] enterprise
    - [-] servers
      - [-] sip-gateway PBX
  - + vsp\session-config-pool
  - + server-pool
  - + sip-gateway Telco
    - dns-group group1
- + accounting
- + dns
  - settings
- + services-routing

### Configuration Loaded

The configuration has been successfully loaded.

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

The screenshot shows the configuration page for a SIP gateway PBX. On the left is a navigation tree under 'Configuration: all' with tabs for 'Configuration', 'Setup', and 'View'. The tree is expanded to 'vsp' > 'enterprise' > 'servers' > 'sip-gateway PBX'. The main content area is titled 'Configure vsp\enterprise\servers\sip-gateway PBX' and includes a 'Show advanced' link. Below the title are buttons for 'Set', 'Reset', 'Back', 'Copy', and 'Delete'. A set of links includes 'Manage connections', 'Log instant messages', 'Record media', 'Record files', 'Set up accounting', 'Change "from:" URI', and 'Change "to:" URI'. The 'general:' section contains a table with the following fields:

general:	
* name	PBX
admin	enabled (Resource is active)
domain	avaya.com
failover-detection	ping (Use OPTIONS to detect failures)

The 'servers:' section contains a table with the following fields:

servers:	
+server-pool	[Delete]

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**.

The screenshot displays the configuration interface for a SIP gateway. On the left, a tree view shows the configuration hierarchy: cluster > vsp > enterprise > servers > sip-gateway Telco. The 'sip-gateway Telco' item is selected and highlighted with a red box. The main panel shows the configuration for this gateway. At the top, there are buttons for 'Set', 'Reset', 'Back', 'Copy', and 'Delete'. Below these are links for 'Manage connections', 'Log instant messages', 'Record media', 'Record files', 'Set up accounting', 'Change "from:" URI', and 'Change "to:" URI'. The configuration is divided into two sections: 'general:' and 'servers:'. The 'general:' section contains the following fields: '\* name' (Telco), 'admin' (enabled), 'domain' (sip.skype.com), and 'failover-detection' (register). The 'servers:' section contains a 'server-pool' entry with a 'Delete' link.

**Configuration: all**

Configuration Setup View

- cluster
  - box:aasbc
- vsp
  - default-session-config
  - pre-session-config
  - session-config-pool
  - dial-plan
  - registration-plan
  - enterprise
    - servers
      - sip-gateway PBX
        - vsp\session-config-pool
        - server-pool
      - sip-gateway Telco
        - vsp\session-config-pool
        - server-pool
    - dns-group group1
  - accounting
  - dns
  - ...

**Configure vsp\enterprise\servers\sip-gateway Telco** Show advanced

Set Reset Back Copy Delete

[Manage connections](#), [Log instant messages](#), [Record media](#), [Record files](#),  
[Set up accounting](#), [Change "from:" URI](#), [Change "to:" URI](#)

**general:**

* name	Telco
admin	enabled (Resource is active)
domain	sip.skype.com
failover-detection	register (Use REGISTER to detect failures)

**servers:**

server-pool	[Delete]
-------------	----------

### 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.

The screenshot shows the acmeApacket Configuration page. The left pane displays a tree view under 'Configuration: all'. The 'vsp' folder is expanded, and 'session-config-pool' is selected. Under 'session-config-pool', 'entry ToTelco' is expanded, and 'to-uri-specification' is highlighted with a red box. The right pane shows 'Configuration Loaded' with the message: 'The configuration has been successfully loaded.'

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

The screenshot shows the acmeApacket Configuration page with the configuration form for 'entry ToTelco to-uri-specification'. The left pane shows the navigation tree with 'to-uri-specification' highlighted. The right pane shows the configuration form with the following fields:

Field	Value	Notes
user	enter to-uri or select from to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
host	enter next-hop-domain or select from next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	enter to-uri or select from to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
display	enter to-uri or select from to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
transport	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
user-param	omit	
user-truncate-non-digits	disabled	(Resource is inactive)
uri-parameter	<a href="#">Add uri-parameter</a>	

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**.

The screenshot shows a web interface with a navigation menu on the left and a configuration form on the right. The left pane shows a tree view under 'vsp' > 'session-config-pool' > 'entry ToTelco' > 'from-uri-specification', which is highlighted with a red box. The right pane is titled 'Configure vsp\session-config-pool\entry ToTelco\from-uri-specification' and contains several fields:

- user**: enter 99051000109702 or select from 99051000109702
- host**: enter next-hop-domain or select from next-hop-domain (Net-Net OS-E uses the domain of the next-hop server.)
- port**: enter from-uri or select from from-uri (Net-Net OS-E uses the value from the incoming FROM URI.)
- display**: enter from-uri or select from from-uri (Net-Net OS-E uses the value from the incoming FROM URI.)
- user-agent-aware-display-translation**: disabled (Resource is inactive)
- transport**: UDP (Net-Net OS-E sets the transport method to UDP.)

Buttons for 'Set', 'Reset', 'Back', and 'Delete' are visible at the top of the form area.

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**.

The screenshot shows a web interface with a navigation menu on the left and a configuration form on the right. The left pane shows a tree view under 'vsp' > 'session-config-pool' > 'entry ToTelco' > 'request-uri-specification', which is highlighted with a red box. The right pane is titled 'Configure vsp\session-config-pool\entry ToTelco\request-uri-specification' and contains several fields:

- user**: enter request-uri or select from request-uri (Net-Net OS-E uses the value from the incoming REQUEST URI.)
- host**: enter next-hop-domain or select from next-hop-domain (Net-Net OS-E uses the domain of the next-hop server.)
- port**: enter request-uri or select from request-uri (Net-Net OS-E uses the value from the incoming REQUEST URI.)
- transport**: request-uri (Net-Net OS-E uses the value from the incoming REQUEST URI.)

Buttons for 'Set', 'Reset', 'Back', and 'Delete' are visible at the top of the form area.

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**.

The screenshot shows a configuration page for 'p-asserted-identity-uri-specification'. On the left, a tree view shows the navigation path: Configuration > vsp > session-config-pool > entry ToTelco > p-asserted-identity-uri-specification. The main area has a title 'Configure vsp\session-config-pool\entry ToTelco\p-asserted-identity-uri-specification' and buttons for 'Set', 'Reset', 'Back', and 'Delete'. Below are three rows of configuration options:

<b>user</b>	enter <input type="text" value="same-uri"/> or select from <input type="text" value="same-uri"/> (Net-Net OS-E uses the value from the uri being altered)
<b>host</b>	enter <input type="text" value="next-hop-domain"/> or select from <input type="text" value="next-hop-domain"/> (Net-Net OS-E uses the domain of the next-hop server.)
<b>port</b>	enter <input type="text" value="same-uri"/> or select from <input type="text" value="same-uri"/> (Net-Net OS-E uses the

## 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.

The screenshot shows the configuration interface for the rule **Configure vsp\session-config-pool\entry ToPBX\to-uri-specification**. On the left, a tree view shows the navigation path: **vsp** > **session-config-pool** > **entry ToPBX** > **to-uri-specification**. The main area has a title bar with **Set**, **Reset**, **Back**, and **Delete** buttons. Below are three rows of configuration options:

Field	Value	Description
user	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
host	next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)

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**.

The screenshot shows the configuration interface for the rule **Configure vsp\session-config-pool\entry ToPBX\from-uri-specification**. On the left, a tree view shows the navigation path: **vsp** > **session-config-pool** > **entry ToPBX** > **from-uri-specification**. The main area has a title bar with **Set**, **Reset**, **Back**, and **Delete** buttons. Below are three rows of configuration options:

Field	Value	Description
user	request-uri	(Net-Net OS-E uses the value from the incoming REQUEST URI.)
host	next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	from-uri	(Net-Net OS-E uses the value from the incoming FROM URI.)

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**.

The screenshot shows the Acme Packet Configuration interface. The left pane shows a tree view with 'request-uri-specification' selected. The main pane displays the configuration for 'Configure vsp\session-config-pool\entry ToPBX\request-uri-specification'. The 'host' field is set to 'next-hop-domain'.

Field	Value	Description
user	request-uri	(Net-Net OS-E uses the value from the incoming REQUEST URI.)
host	next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	request-uri	(Net-Net OS-E uses the value from the incoming REQUEST URI.)

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**.

The screenshot shows the Acme Packet Configuration interface. The left pane shows a tree view with 'p-asserted-identity-uri-specification' selected. The main pane displays the configuration for 'Configure vsp\session-config-pool\entry ToPBX\p-asserted-identity-uri-specification'. The 'host' field is set to 'next-hop-domain'.

Field	Value	Description
user	request-uri	(Net-Net OS-E uses the value from the incoming REQUEST URI.)
host	next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	same-uri	(Net-Net OS-E uses the value from the incoming uri being altered.)

## 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**.

The screenshot shows the Avaya Aura configuration interface. On the left, a tree view under 'Configuration: all' shows the path: cluster > vsp > pre-session-config > sip-header-settings. The 'sip-header-settings' item is highlighted. On the right, the 'Configure vsp\pre-session-config\sip-header-settings' page is displayed. It features buttons for 'Set', 'Reset', 'Back', and 'Delete'. Below these, the 'admin' status is 'enabled' (Resource is active). The 'rule' section has an 'Add rule' button highlighted with a red box. At the bottom, there are 'Set', 'Reset', and 'Back' buttons, and links for 'Help' and 'Index'.

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

The screenshot shows the Avaya Aura configuration interface. On the left, the tree view shows the path: cluster > vsp > pre-session-config > sip-header-settings. On the right, the 'Create vsp\pre-session-config\sip-header-settings\rule - Step 1 of 1: Edit rule' dialog box is displayed. It contains the instruction: 'Please provide some basic information for rule. Then press "Create".' There is a text input field for '\* name' with 'P-Site' entered. Below the input field are buttons for 'Create', 'Reset', and 'Cancel'. The 'Create' button is highlighted with a red box.

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**.

**Configuration: all**

Configuration Setup View

- cluster
  - box:aasbc
- vsp
  - default-session-config
  - pre-session-config
    - sip-header-settings
      - rule P-Site
  - session-config-pool
  - dial-plan
    - registration-plan
  - enterprise
  - accounting
  - dns
    - settings
  - services-routing

**Configure vsp\pre-session-config\sip-header-settings\rule P-Site** [Help](#) [Index](#)

Set Reset Back Copy Delete

\* name P-Site

description

condition

\* condition-type match-header (Sets the name of the header to match on)

\* name enter P-Site or select from P-Site

action **Configure**

Set Reset Back Copy

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**.

**Configuration: all**

Configuration Setup View

- cluster
  - box:aasbc
- vsp
  - default-session-config
  - pre-session-config
    - sip-header-settings
      - rule P-Site
    - session-config-pool
    - dial-plan

**Create vsp\pre-session-config\sip-header-settings\rule P-Site\action** [Index](#)

Please provide some basic information for action. Then press "Create".

\* action-type strip-header (The Net-Net OS-E removes the SIP head)

**Create** Reset Cancel

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

**Configuration: all**

Configuration Setup View

- [-] cluster
  - [-] box:aasbc
- [-] vsp
  - [-] default-session-config
  - [-] pre-session-config
    - [-] sip-header-settings
      - rule P-Site
  - [-] session-config-pool
  - [-] dial-plan
    - registration-plan
  - [-] enterprise
  - [-] accounting
  - [-] dns
    - settings
  - [-] services-routing

**Configure vsp\pre-session-config\sip-header-settings\rule P-Site** [Help](#) [Index](#)

Set Reset Back Copy Delete

<b>* name</b>	<input type="text" value="P-Site"/>
<b>description</b>	<input type="text"/>
<b>condition</b>	<b>* condition-type</b> <input type="text" value="match-header"/> (Sets the name of the header to match) <b>* name</b> enter <input type="text" value="P-Site"/> or select from <input type="text" value="P-Site"/>
<b>action</b>	<b>* action-type</b> <input type="text" value="strip-header"/> (The Net-Net OS-E removes the SIP header)

Set Reset Back Copy

[Help](#) [Index](#)

## 7.6. Save the Configuration

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

The screenshot shows the configuration interface. On the left, under 'Configuration: all', the 'Configuration' tab is selected, and the 'Update and save configuration' option is highlighted in a blue box. Below it, a dropdown menu is visible with options: 'Reload configuration', 'Validate configuration', 'Analyze configuration', 'Search configuration', 'Save as XML', and 'Load from XML'. The main area shows the configuration for 'rule P-Site' with fields for name, description, condition, and action. The condition is set to 'match-header' and the action is 'strip-header'.

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

The screenshot shows the configuration interface after the update. A message box at the top reads 'Configuration Updated and Saved' with the text 'The running configuration has been updated and saved.' below it. The configuration details for 'rule P-Site' are visible, showing the same settings as in the previous screenshot.

## 8. General Test Approach and Test Results

The test environment consisted of:

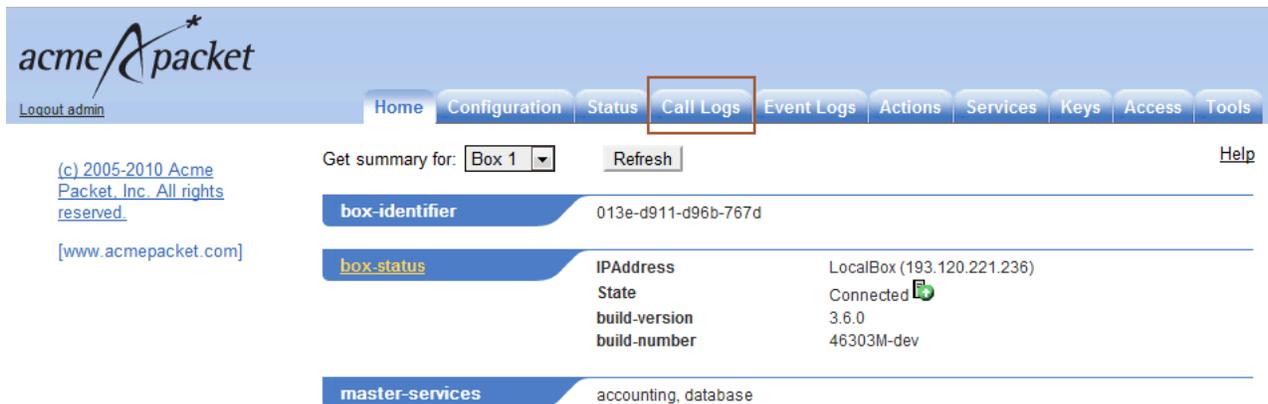
- A simulated enterprise with Avaya Aura™ System Manager, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager, Avaya phones, Avaya Aura™ Session Border Controller, and Avaya Modular Messaging.
- A production version of the Skype Connect service, to which the simulated enterprise was connected.

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 screenshot shows the Avaya Aura Session Border Controller GUI. The top navigation bar includes links for Home, Configuration, Status, Call Logs (highlighted with a red box), Event Logs, Actions, Services, Keys, Access, and Tools. Below the navigation bar, there is a section for 'Get summary for: Box 1' with a 'Refresh' button and a 'Help' link. The main content area displays details for 'box-identifier' (013e-d911-d96b-767d) and 'box-status' (IPAddress: LocalBox (193.120.221.236), State: Connected, build-version: 3.6.0, build-number: 46303M-dev). A 'master-services' section at the bottom lists 'accounting, database'.

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).

**acme packet** Call Logs

Status Summary Logout admin Home Configuration Status **Call Logs** Event Logs Actions Services Keys Access Tools

Select:

- Sessions
- User Sessions
- Devices
- SIP Messages
- H323 Messages
- Accounting Calls
- Monitored URIs
- Monitored Calls
- Files
- Database Archives

**Sessions** seconds Refresh

Search Type: All Sessions

Page 1 of 1 showing 30 items View: User Messages

Created	Method	Result	From	To	Call ID
12:39:58.780 Mon 2010-06-14	INVITE		sip:+12024702967@avaya.com	sip:0035312075630@avaya.com	8036baec1f81df141504c0f5cd00

Page 1 of 1 showing 30 items

Taken Jun 14, 2010 12:40:45 PM

## 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.

### Get a Skype button

Skype buttons can be used on your website, blog or even in your email signature to let other people contact you easily.

You can choose from the simple options below, or customise the colours, functions and styles using our [Skype buttons wizard](#).

#### Enter your Skype Name

#### Select a button from below

- 
- 
- 
- 

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.

### Preview your button



### Copy & paste this code

Show  Web HTML  Email HTML

[Save this html snippet](#) to your computer.

```
<!--  
Skype 'Skype Me™!' button  
http://www.skype.com/go/s  
kypebuttons  
-->  
<script  
type="text/javascript"  
src="http://download.skyp  
e.com/share/skypebuttons/  
js/skypeCheck.js"></scrip
```

---

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