



Avaya Solution & Interoperability Test Lab

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

Abstract

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

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

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

Table of Contents

1.	Introduction.....	4
1.1.	Design Principles and Assumptions	5
1.2.	Interoperability Compliance Testing.....	5
1.3.	Abbreviations	5
1.4.	Known Limitations	6
1.5.	Reference Configuration	7
1.6.	Support	8
2.	Equipment and Software Validated	8
3.	Call Flows	9
3.1.	Inbound from PSTN	9
3.2.	Outbound to PSTN	10
3.3.	Inbound from Skype Client	11
3.4.	Skype Connect to Skype P2P Network	12
4.	Configure Avaya Aura™ Session Manager.....	13
4.1.	Log in to Avaya Aura™ System Manager using the GUI.....	13
4.2.	Administer SIP Domain	14
4.3.	Define a Location	15
4.4.	Define SIP Entities	17
4.5.	Define Entity Links	20
4.6.	Define Routing Policies.....	23
4.7.	Define Dial Patterns	26
5.	Avaya Aura™ Communication Manager	30
5.1.	Administer System Parameters	30
5.2.	Administer Dial Plan.....	31
5.3.	Administer IP Network Parameters.....	32
5.4.	Administer Signaling Group	34
5.5.	Administer Trunk Group	35
5.6.	Administer Automatic Route Selection.....	36
5.7.	Administer Route Pattern	37
5.8.	Administer Public Unknown Numbering.....	37
5.9.	Administer Incoming Call Handling Treatment.....	38
6.	Configure Skype Manager	39
6.1.	Log in to Skype Manager	39
6.2.	Create SIP Profile	40
6.3.	Administer Authentication Method.....	43
6.4.	Administer Maximum Simultaneous Calls	46
6.5.	Administer Online Numbers	50
6.6.	Administer Caller ID	56
6.7.	Create a Business User	60
6.8.	Add Business User to SIP Profile.....	64
7.	Configure Avaya Aura™ Session Border Controller.....	69
7.1.	Log in to Avaya Aura™ Session Border Controller using the GUI.....	69
7.2.	Administer SIP Domains	70

7.3.	Administer Outbound SIP Header Manipulation Rules	74
7.4.	Administer Inbound SIP Header Manipulation Rules.....	77
7.5.	Administer SIP Header Rules for Topology Hiding	79
7.6.	Save the Configuration	83
8.	General Test Approach and Test Results	84
9.	Verification Steps.....	84
9.1.	Troubleshooting Tools	85
10.	Conclusion	85
11.	References	86
Appendix A		87

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.

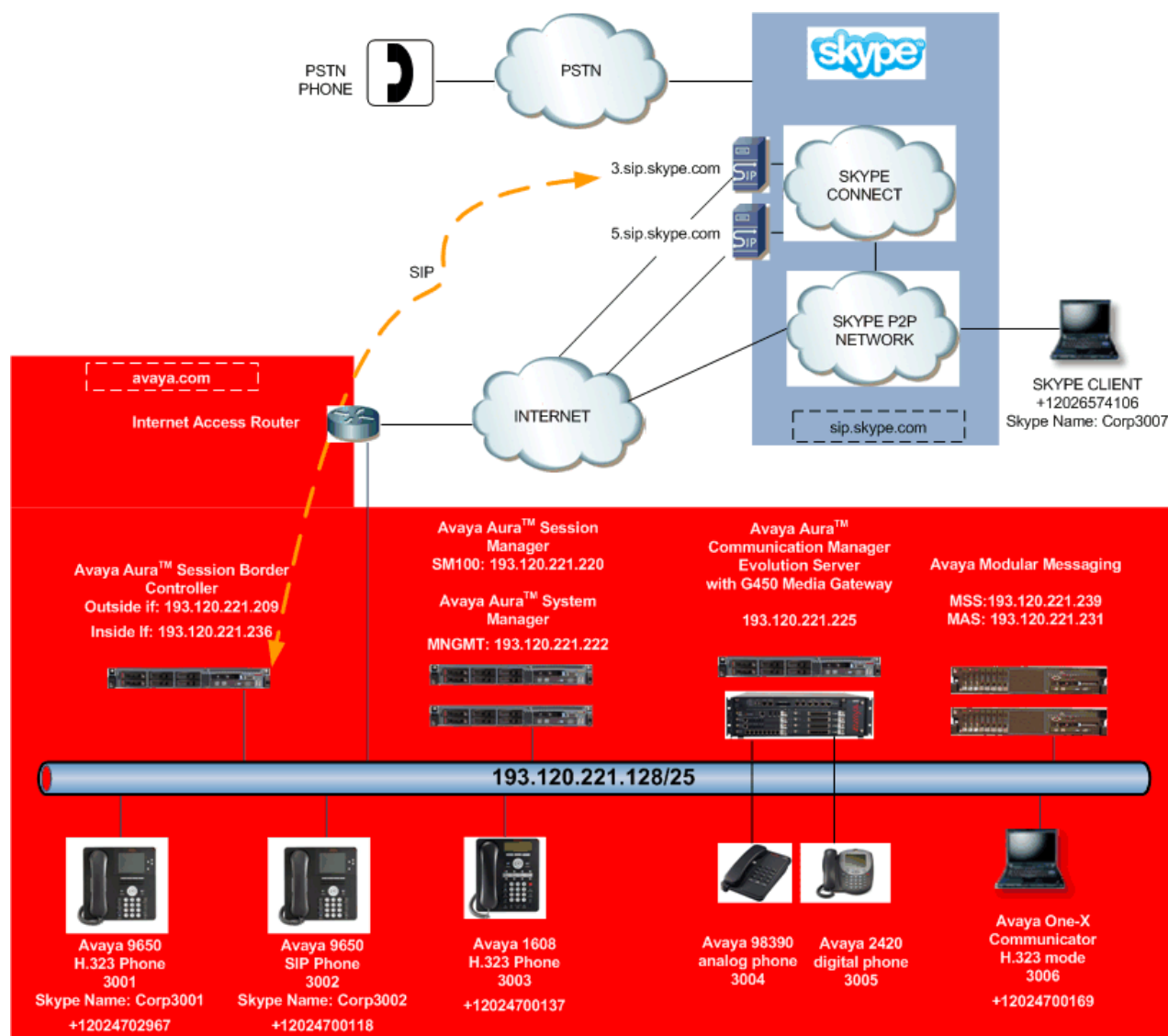


Figure 1: Avaya Interoperability Test Lab Reference Configuration

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.

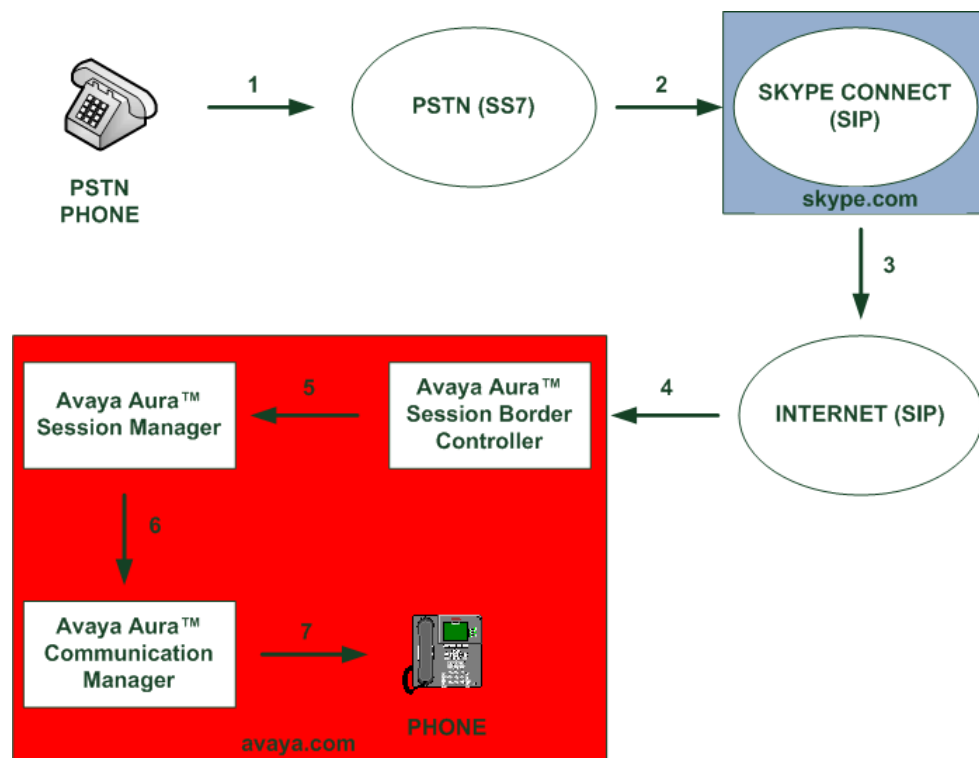


Figure 2 - Inbound PSTN to Skype Connect Call

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.

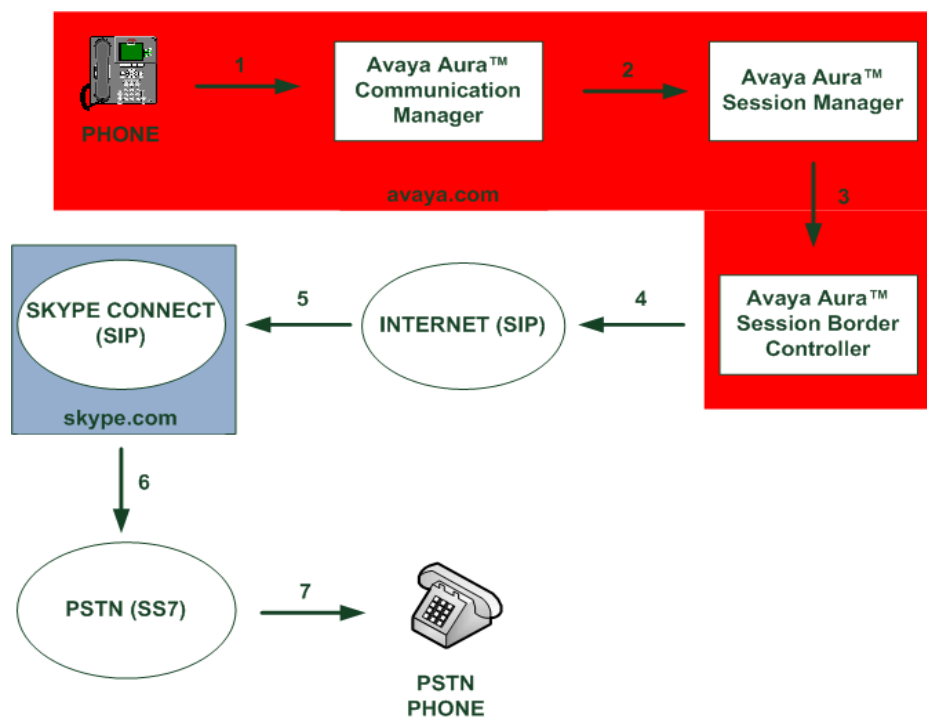


Figure 3: Outbound Skype Connect to PSTN call with CLIP

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.

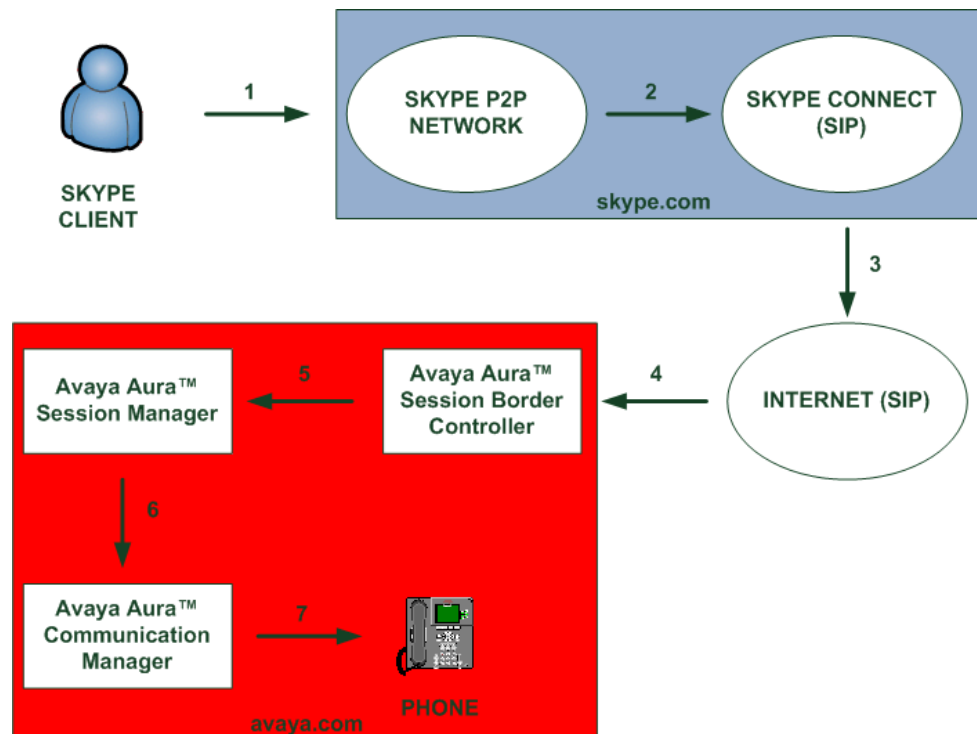


Figure 4: Inbound Skype P2P to Skype Connect call

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.

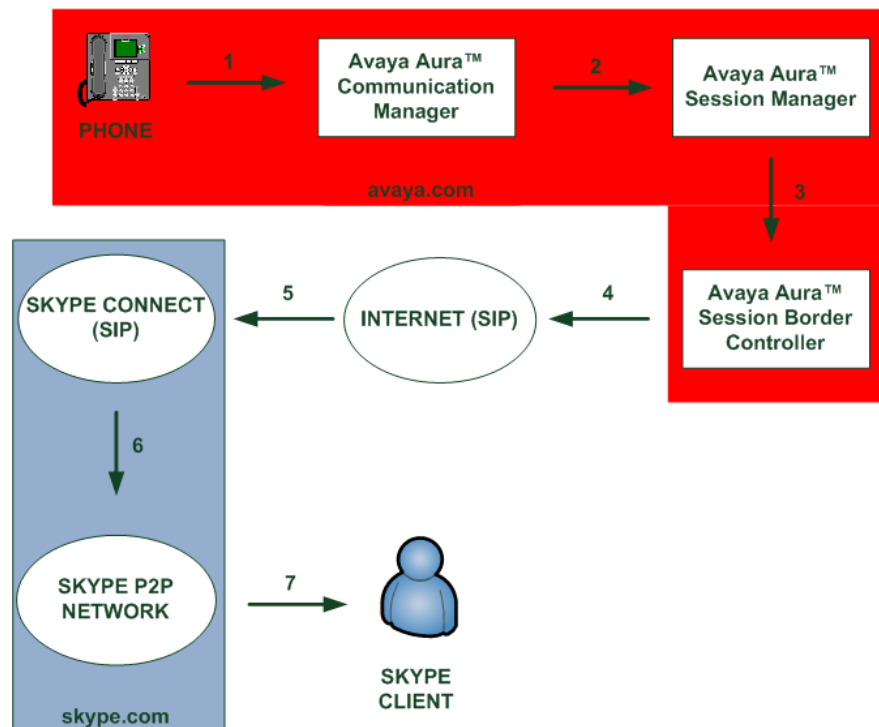


Figure 5: Outbound Skype Connect to Skype P2P Client call

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**.



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

▸ 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

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**.



Avaya Aura™ System Manager
6.0

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Home / Routing / Domains

▸ Elements

▸ Events

▸ Groups & Roles

Licenses


▼ Routing

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 Aura™ System Manager
6.0

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[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Domains

▶ Elements

▶ Events

▶ Groups & Roles

Licenses

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

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 Aura™ System Manager
6.0

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[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Locations

▶ Elements

▶ Events

▶ Groups & Roles

Licenses

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

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

▸ 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

Location Details

CommitCancel

General

* Name: enterprise

Notes:

Managed Bandwidth: Kbit/sec

* Average Bandwidth per Call: 80 Kbit/sec

Location Pattern

AddRemove

1 Item RefreshFilter: Enable

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

Select : All, None

* Input Required


CommitCancel

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.



Avaya Aura™ System Manager
6.0

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2010 1:22 PM
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Home / Routing / SIP Entities

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

SIP Entities

Edit

New

Duplicate

Delete

More Actions ▼

Commit

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

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Home / Routing / SIP Entities / SIP Entity Details

SIP Entity Details Commit Cancel

General

* **Name:** cmes

* **FQDN or IP Address:** 193.120.221.225

Type: CM

Notes: CM - Evolution Server R6.0

Adaptation:

Location: enterprise

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

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**.

The screenshot displays the Avaya Aura™ System Manager 6.0 web interface. The top navigation bar includes 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'. Below this is a red breadcrumb trail: 'Home / Routing / SIP Entities / SIP Entity Details'. On the left is a sidebar menu with categories like Elements, Events, Groups & Roles, Licenses, Routing (expanded), Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, Defaults, Security, System Manager Data, and Users. The main content area is titled 'SIP Entity Details' and has a 'General' tab selected. A red box highlights the 'Name' field (value: 'aasbc'), 'FQDN or IP Address' field (value: '193.120.221.236'), 'Type' dropdown (value: 'SIP Trunk'), and 'Notes' field (value: 'AASBC - inside if'). Below this, the 'Adaptation' dropdown is empty, the 'Location' dropdown is set to 'enterprise' (highlighted with a red box), and the 'Time Zone' dropdown is set to 'Etc/GMT'. There is an unchecked checkbox for 'Override Port & Transport with DNS SRV:'. The '* SIP Timer B/F (in seconds):' field is set to '4'. The 'Credential name' field is empty. The 'Call Detail Recording' dropdown is set to 'egress'. In the top right corner of the form area, the 'Commit' button is highlighted with a red box, next to a 'Cancel' button.

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**.



Avaya Aura™ System Manager
6.0

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[Help](#) | [About](#) | [Change Password](#) | [Log off](#)

Home / Routing / Entity Links

▸ Elements
▸ Events
▸ Groups & Roles
Licenses
▼ Routing
 Domains
 Locations
 Adaptations
 SIP Entities
 Entity Links
 Time Ranges

Entity Links

Edit **New** Duplicate Delete More Actions ▾ Commit

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

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[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**.



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

▶ Elements

▶ Events

▶ Groups & Roles

Licenses

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links


Entity Links

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**.



Avaya Aura™ System Manager 6.0 Web 3.0

Home / Routing / Routing Policies

▶ Elements

▶ Events

▶ Groups & Roles

Licenses

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Routing Policies

Edit

New

Duplicate

Delete

More Actions ▼

Commit

Items Refresh

<input type="checkbox"/>	Name	Disabled	Destination
<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

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

Commit

Cancel

General

* Name:

toCMES-30xx

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 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"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

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

Help

Help for Routing Policy Details fields

Routing Policy Details

Commit Cancel

General

* Name: toSkype-00

Disabled: ☐

Notes:

SIP Entity as Destination

Select

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

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

	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"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

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

[Edit](#) [New](#) [Duplicate](#) [Delete](#) [More Actions](#) [Commit](#)

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

	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	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

	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination
<input type="checkbox"/>	enterprise		toCMES-30xx	0	<input type="checkbox"/>	cmes

Select : All, None

Help

Help for Dial Pattern Details fields

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

1 Item Refresh Filter: Enable

	Originating Location Name ¹	Originating Location Notes	Routing Policy Name	Rank ²	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	
Maximum Administered SIP Trunks:	5000	250	
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0	
Maximum Number of DS1 Boards with Echo Cancellation:	0	0	
Maximum TN2501 VAL Boards:	10	1	
Maximum Media Gateway VAL Sources:	0	0	
Maximum TN2602 Boards with 80 VoIP Channels:	128	0	
Maximum TN2602 Boards with 320 VoIP Channels:	128	2	
Maximum Number of Expanded Meet-me Conference Ports:	0	0	
(NOTE: You must logoff & login to effect the permission changes.)			

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

display system-parameters customer-options		Page 4 of 11
OPTIONAL FEATURES		
Emergency Access to Attendant? Y	IP Stations? Y	
Enable 'dadmin' Login? Y		
Enhanced Conferencing? Y	ISDN Feature Plus? Y	
Enhanced EC500? Y	ISDN/SIP Network Call Redirection? N	
Enterprise Survivable Server? N	ISDN-BRI Trunks? Y	
Enterprise Wide Licensing? N	ISDN-PRI? Y	
ESS Administration? N	Local Survivable Processor? N	
Extended Cvg/Fwd Admin? Y	Malicious Call Trace? N	
External Device Alarm Admin? N	Media Encryption Over IP? N	
Five Port Networks Max Per MCC? N	Mode Code for Centralized Voice Mail? N	
Flexible Billing? N		
Forced Entry of Account Codes? N	Multifrequency Signaling? Y	
Global Call Classification? N	Multimedia Call Handling (Basic)? Y	
Hospitality (Basic)? Y	Multimedia Call Handling (Enhanced)? Y	
Hospitality (G3V3 Enhancements)? N	Multimedia IP SIP Trunking? N	
IP Trunks? Y		
IP Attendant Consoles? N		

5.2. Administer Dial Plan

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

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

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

5.3. Administer IP Network Parameters

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

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

Change ip-codec-set 2		Page 1 of 2	
IP Codec Set			
Codec Set: 2			
Audio 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	
FAX	off	0	
Modem	off	0	
TDD/TTY	off	0	
Clear-channel	n	0	

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

Change ip-network-region 1		Page 1 of 19
IP NETWORK REGION		
Region: 1		
Location: Authoritative Domain: avaya.com		
Name: Skype		
MEDIA PARAMETERS		
Intra-region IP-IP Direct Audio: yes		
Inter-region IP-IP Direct Audio: yes		
IP Audio Hairpinning? N		
Codec Set: 1		
UDP Port Min: 2048		
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		
Inter-region IP-IP Direct Audio: yes		
IP Audio Hairpinning? N		
Codec Set: 2		
UDP Port Min: 2048		
UDP Port Max: 3329		
DIFFSERV/TOS PARAMETERS		
RTCP Reporting Enabled? Y		
Call Control PHB Value: 46		
RTCP MONITOR SERVER PARAMETERS		
Audio PHB Value: 46		
Use Default Server Parameters? Y		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
AUDIO RESOURCE RESERVATION PARAMETERS		

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

change ip-network-region 2		Page 4 of 19
Source Region: 2		Inter Network Region Connection Management
		I M
		G A e
dst	codec direct	WAN-BW-limits Video Intervening Dyn A G a
rgn	set WAN Units Total Norm Prio Shr Regions	CAC R L s
1	2 y NoLimit	n

5.4. Administer Signaling Group

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

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

```
Add signaling-group 100
```

```
SIGNALING GROUP
```

```
Group Number: 100
```

```
Group Type: sip
```

```
Transport Method: tcp
```

```
IMS Enabled? N
```

```
Near-end Node Name: procr
```

```
Far-end Node Name: sm100
```

```
Near-end Listen Port: 5060
```

```
Far-end Listen Port: 5060
```

```
Far-end Network Region:2
```

```
Far-end Domain:
```

```
Bypass If IP Threshold Exceeded? n
```

```
Incoming Dialog Loopbacks: eliminate
```

```
RFC 3389 Comfort Noise? n
```

```
DTMF over IP: rtp-payload
```

```
Direct IP-IP Audio Connections? y
```

```
Session Establishment Timer(min): 3
```

```
IP Audio Hairpinning? n
```

```
Enable Layer 3 Test? Y
```

```
Direct IP-IP Early Media? n
```

```
H.323 Station Outgoing Direct Media? n
```

```
Alternate Route Timer(sec): 6
```

5.5. Administer Trunk Group

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

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

Add trunk-group 100	Page 1 of 21		
TRUNK GROUP			
Group Number: 100	Group Type: sip	CDR Reports: y	
Group Name: SkypeConnect	COR: 1	TN: 1	TAC: *22
Direction: both	Outgoing Display? n		
Dial Access? n		Night Service:	
	Auth Code? n		
Service Type: public-ntwrk			
	Signaling Group: 52		
	Number of Members: 20		

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

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

add trunk-group 100	Page 3 of 21		
TRUNK FEATURES			
ACA Assignment? n	Measured: none		
	Maintenance Tests? y		
Numbering Format: public			
	UI 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 String	Total		Route Pattern	Call Type	Node Num	ANI	
	Min	Max				Reqd	
011	10	18	100	intl		n	

5.7. Administer Route Pattern

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

change route-pattern 100										Page 1 of 3		
Pattern Number: 68 Pattern Name:												
SCCAN? n Secure SIP? n												
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted			DCS/	IXC	
No			Mrk	Lmt	List	Del	Digits			QSIG		
Dgts										Intw		
1:	100	0									n	user

5.8. Administer Public Unknown Numbering

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

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

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

5.9. Administer Incoming Call Handling Treatment

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

change inc-call-handling-trmt trunk-group 100					Page	1	of	30
INCOMING CALL HANDLING TREATMENT								
Service/ 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 (Avaya, Account details, tony.skype11, Sign out, Help, Chat support) are on the right. Below this is a dark toolbar with icons for Dashboard, Features (highlighted with a red box), and Reports. The current balance is €226,94, and there is a 'Buy Skype Credit' button. A search bar for members is also present. The main content area is divided into two sections: 'Reports' and 'Your account'. The 'Reports' section includes an 'Allocations' bar chart showing credit 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 columns: 'Your features' (showing 6 members with Skype Credit), 'Your Members' (showing 13 members and an 'Add Members' button), and 'News' (a welcome message about Skype Manager).

skype manager™

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

Dashboard Features Reports

€226,94 Buy Skype Credit Search Members

Reports Your account Account status: All OK ✓

Allocations

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.

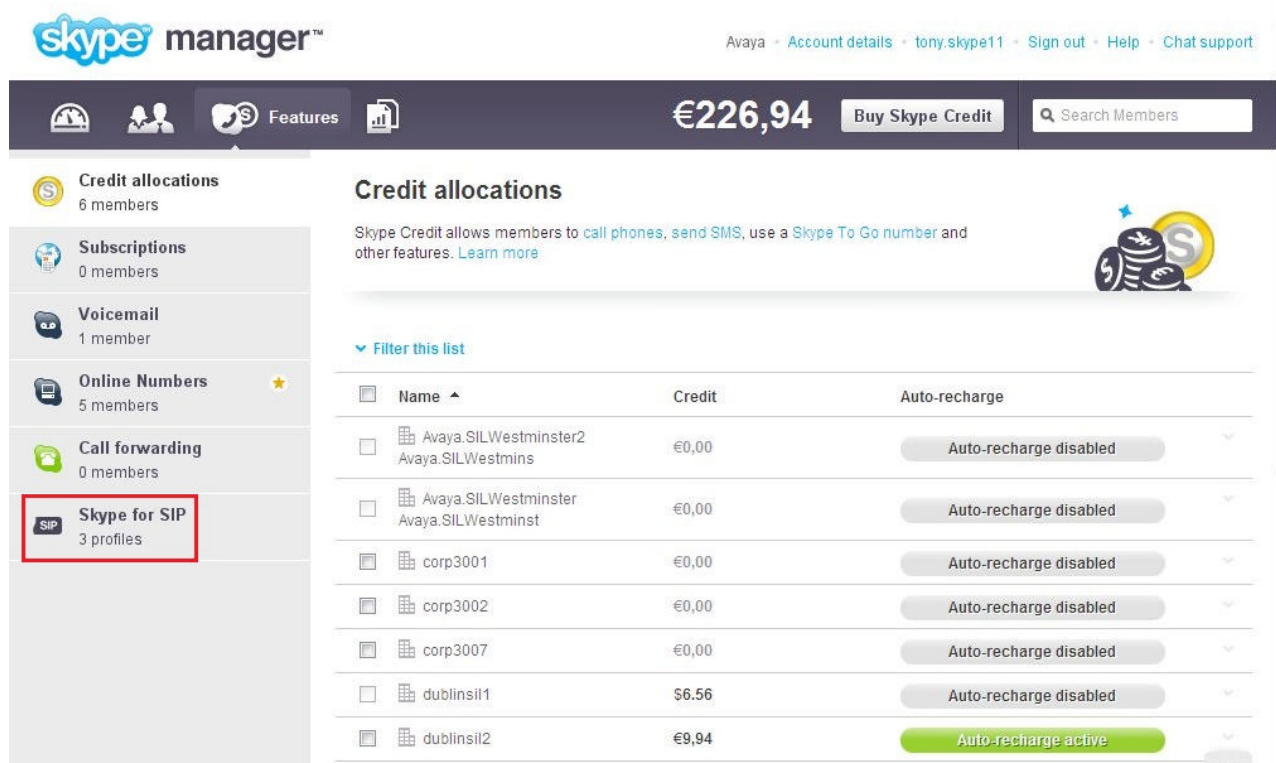
Your features Your Members News

6 members have Skype Credit

Your Skype Manager has 13 members Add Members

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.



skype manager™

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

€226,94 Buy Skype Credit Search Members

Features

- Credit allocations 6 members
- Subscriptions 0 members
- Voicemail 1 member
- Online Numbers 5 members
- Call forwarding 0 members
- Skype for SIP 3 profiles**

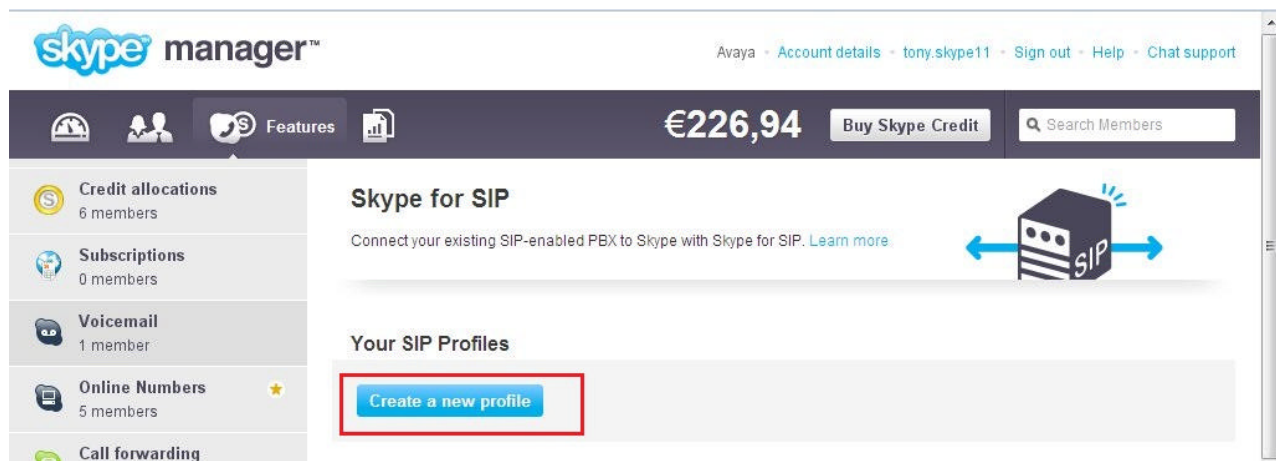
Credit allocations

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**.



skype manager™

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

€226,94 Buy Skype Credit Search Members

Features

- Credit allocations 6 members
- Subscriptions 0 members
- Voicemail 1 member
- Online Numbers 5 members
- Call forwarding

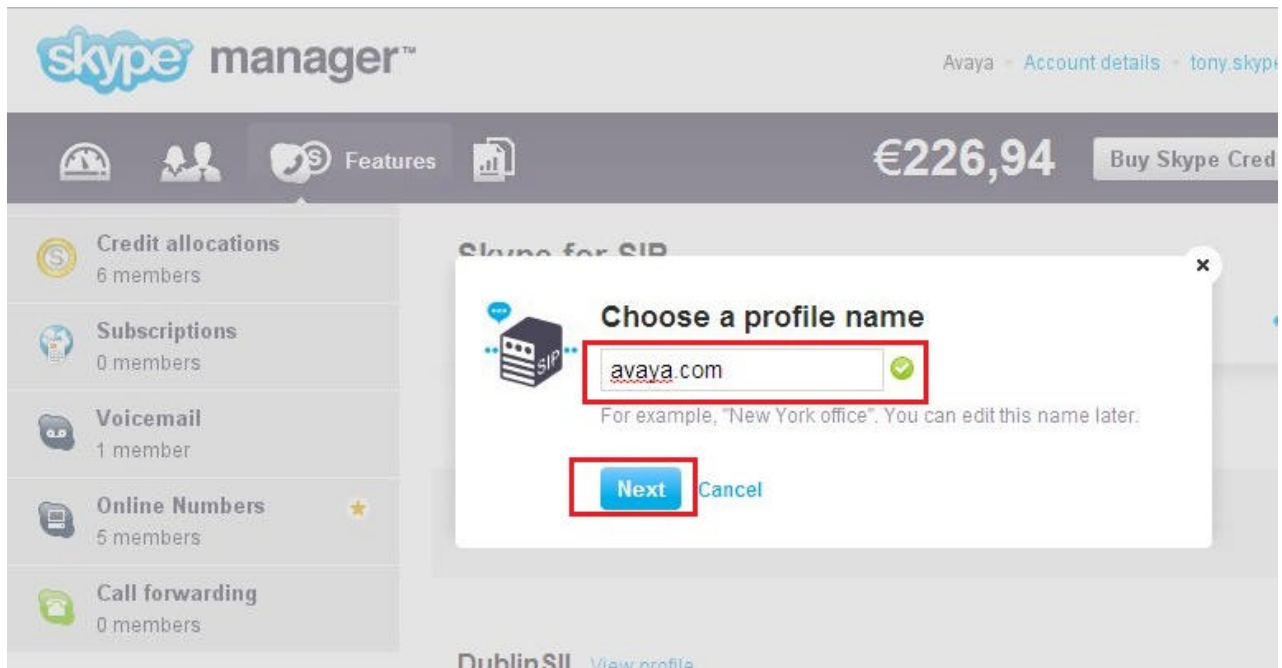
Skype for SIP

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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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 Generate a new password
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**.

The screenshot shows the Skype Manager interface. At the top, the 'skype manager' logo is on the left, and 'Avaya · Account details · tony.skype11' is on the right. Below the logo is a dark navigation bar with icons for a clock, people, a speech bubble with a dollar sign, and a bar chart. The text 'Features' is next to the speech bubble icon. On the right of this bar, the balance '€226,94' is displayed, along with a 'Buy Skype Credit' button. The left sidebar contains a 'PBX' icon, the domain 'avaya.com', and links for 'Profile settings', 'Authentication details' (which is highlighted), and 'Reports'. At the bottom of the sidebar is a link '« Back to SIP Profile list'. The main content area is titled 'Authentication details'. It prompts the user to 'Please choose the method of authentication needed for your PBX.' There are two options: 'Registration (Username/password)' with a green checkmark, and 'or, IP Authentication' with a blue question mark icon. The 'IP Authentication' option is selected. Below this, it says 'Please enter the IP details for your PBX'. There are two input fields: 'Public IP address' with the value '193.120.221.209' and 'UDP Port' with the value '5060'. A blue 'Continue' button is at the bottom right of the form. Red boxes highlight the 'Public IP address' field, the 'UDP Port' field, and the 'Continue' button.

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Avaya · Account details · tony.skype11

Features

€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 'Profile settings' page for an 'avaya.com' profile. The left sidebar contains navigation links: '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, the 'Your PBX details' section lists the SIP User as '99051000109702', the Public IP address as '193.120.221.209', and the UDP Port as '5060'. A 'Change PBX details' link is provided. The 'Use these details to configure your PBX' section shows 'Skype for SIP addresses' with a Primary address of '3.sip.skype.com' and a Secondary address of '5.sip.skype.com'. It also lists 'Skype for SIP IP addresses' with a Primary address of '193.120.218.68' and a Secondary address of '78.141.179.70', with a note to 'enable traffic for these IP addresses in your firewall'.

Your PBX details	
SIP User	99051000109702
Public IP address	193.120.221.209
UDP Port	5060
Change PBX details	





Use these details to configure your PBX	
Skype for SIP addresses	
Primary	3.sip.skype.com
Secondary	5.sip.skype.com
Skype for SIP IP addresses enable traffic for these IP addresses in your firewall	
Primary	193.120.218.68
Secondary	78.141.179.70

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, the Skype Manager logo is on the left, and navigation links (Avaya, Account details, tony.skype11, Sign out, Help, Chat support) are on the right. Below the header is a dark bar containing a balance of €226,94, a 'Buy Skype Credit' button, and a 'Search Members' search bar. The main content area is titled 'Profile settings' and features a sidebar on the left with links for 'Profile settings', 'Authentication details', and 'Reports'. The 'Profile settings' section includes a profile icon for 'avaya.com' and a list of settings: 'Profile name' (avaya.com), 'Calling channels' (with a red box around the link 'Buy a channel subscription to activate this profile'), 'Outgoing calls' (Set up outgoing calls), 'Caller ID' (Set up Caller ID), and 'Incoming calls' (Add a number or business account). A 'Delete this profile' button is located at the bottom of the settings list.

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

 Features 

€226,94 [Buy](#)


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 web interface. At the top, the 'skype manager' logo is on the left, and user information 'Avaya · Account details · tony.skype11 · Sign out · Help · Chat support' is on the right. Below the header is a dark navigation bar with icons for home, contacts, features, and a balance display of '€212,09'. A 'Buy Skype Credit' button and a 'Search Members' search bar are also present. The left sidebar contains a list of services: 'Credit allocations' (6 members), 'Subscriptions' (0 members), 'Voicemail' (1 member), 'Online Numbers' (5 members, marked with a star), '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 diagram of a SIP server. A green confirmation message box states: 'Subscription changed successfully for Profile avaya.com. The profile's updated subscription setting will be available in a few minutes.' Below this is a section titled 'Your SIP Profiles' with a 'Create a new profile' button. Under the 'avaya.com' profile, there is a table with settings: 'Channels' (link: 'Buy a channel subscription to activate this profile'), 'Outgoing calls' (link: 'Set up outgoing calls', highlighted with a red box), and 'Incoming calls' (link: 'Set up incoming calls').

skype manager™

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

€212,09 Buy Skype Credit Search Members

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](#)

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](#)

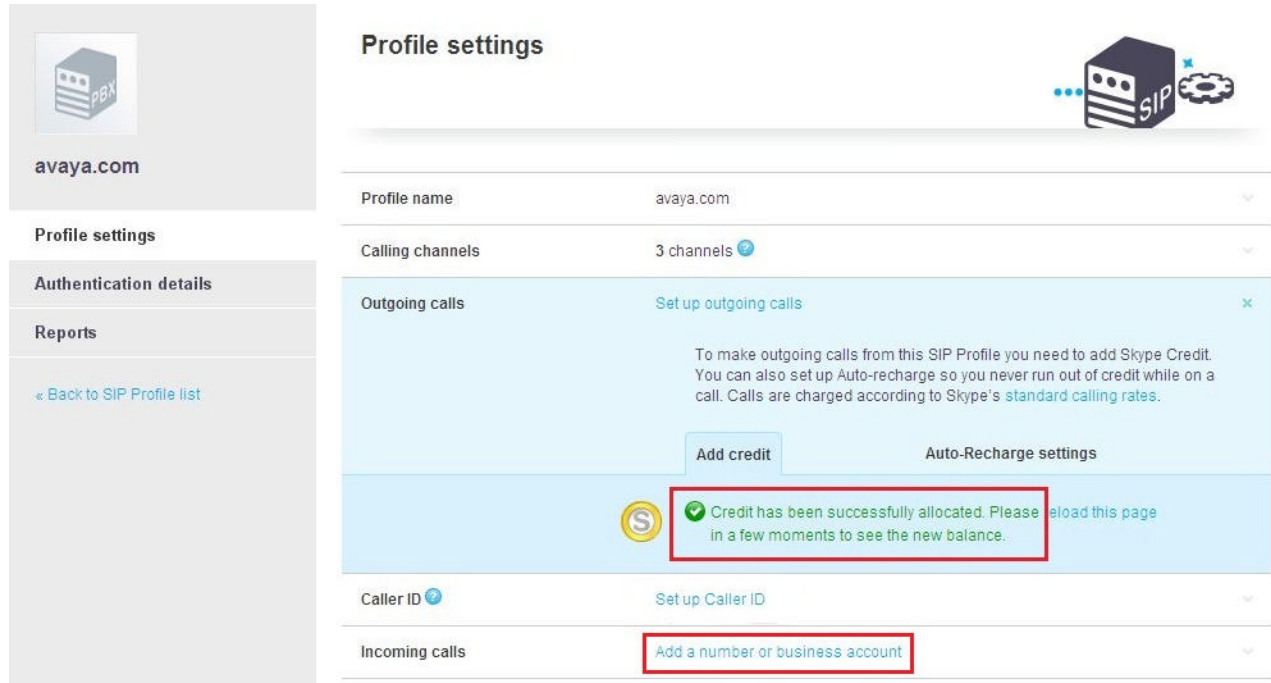
Channels	Buy a channel subscription to activate this profile
Outgoing calls	Set up outgoing calls
Incoming calls	Set up incoming calls

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

The screenshot shows the Skype Manager interface. 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 header is a dark navigation bar with icons for a dashboard, users, features, and a balance of €212,09. A 'Buy Skype Credit' button and a 'Search Members' search bar are also present. The left sidebar contains a profile icon for 'avaya.com' and a menu with 'Profile settings' (selected), 'Authentication details', and 'Reports'. A link to 'Back to SIP Profile list' is at the bottom of the sidebar. The main content area is titled 'Profile settings' and includes a SIP icon. It displays 'Profile name' as 'avaya.com' and 'Calling channels' as '3 channels'. The 'Outgoing calls' section has a 'Set up outgoing calls' link and a message about adding Skype Credit. Below this, there are tabs for 'Add credit' and 'Auto-Recharge settings'. The 'Add credit' tab is active, showing a currency selector (€), an input field with '10.00', and a blue 'Add credit' button, all of which 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 links: 'Profile settings', 'Authentication details', 'Reports', and a link to 'Back to SIP Profile list'. The main content area shows settings for 'Calling channels' (3 channels), 'Outgoing calls' (with a link to 'Set up outgoing calls'), 'Caller ID' (with a link to 'Set up Caller ID'), and 'Incoming calls' (with a link to 'Add a number or business account'). A large blue banner at the bottom of the 'Outgoing calls' section contains a green checkmark icon and the message: 'Credit has been successfully allocated. Please reload this page in a few moments to see the new balance.' The banner also includes buttons for 'Add credit' and 'Auto-Recharge settings'.

Profile settings

avaya.com

Profile settings

Authentication details

Reports

[Back to SIP Profile list](#)

Profile name: avaya.com

Calling channels: 3 channels

Outgoing calls: [Set up outgoing calls](#)

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](#).

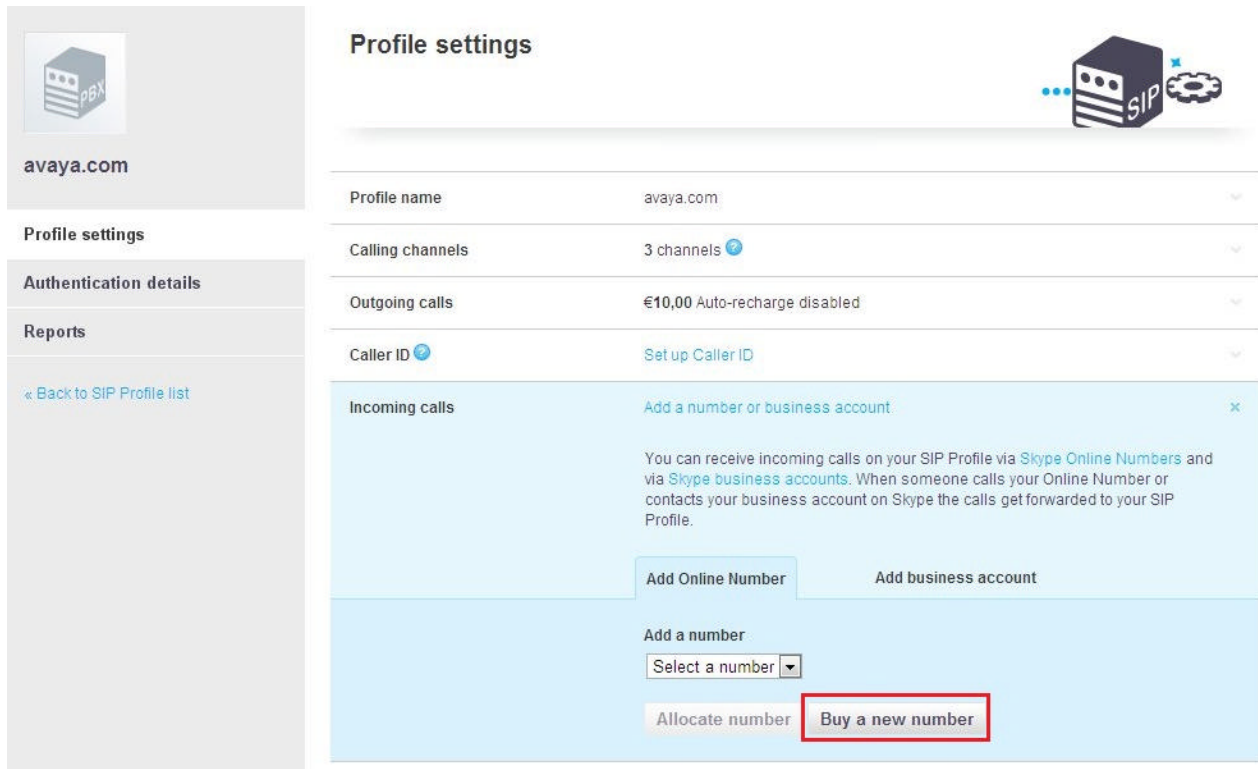
[Add credit](#) [Auto-Recharge settings](#)

✓ Credit has been successfully allocated. Please reload this page in a few moments to see the new balance.

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 links: 'avaya.com', 'Profile settings' (selected), 'Authentication details', 'Reports', and a link to '« Back to SIP Profile list'. The main content area is titled 'Profile settings' and features a SIP icon. Below this is a table of settings:

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

The 'Incoming calls' tab is expanded, displaying the following information:

- Tab title: Incoming calls
- Header: Add a number or business account
- Text: 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.
- Buttons: Add Online Number, Add business account
- Section: Add a number
- Dropdown: Select a number
- Buttons: Allocate number, Buy a new number (highlighted with a red box)

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

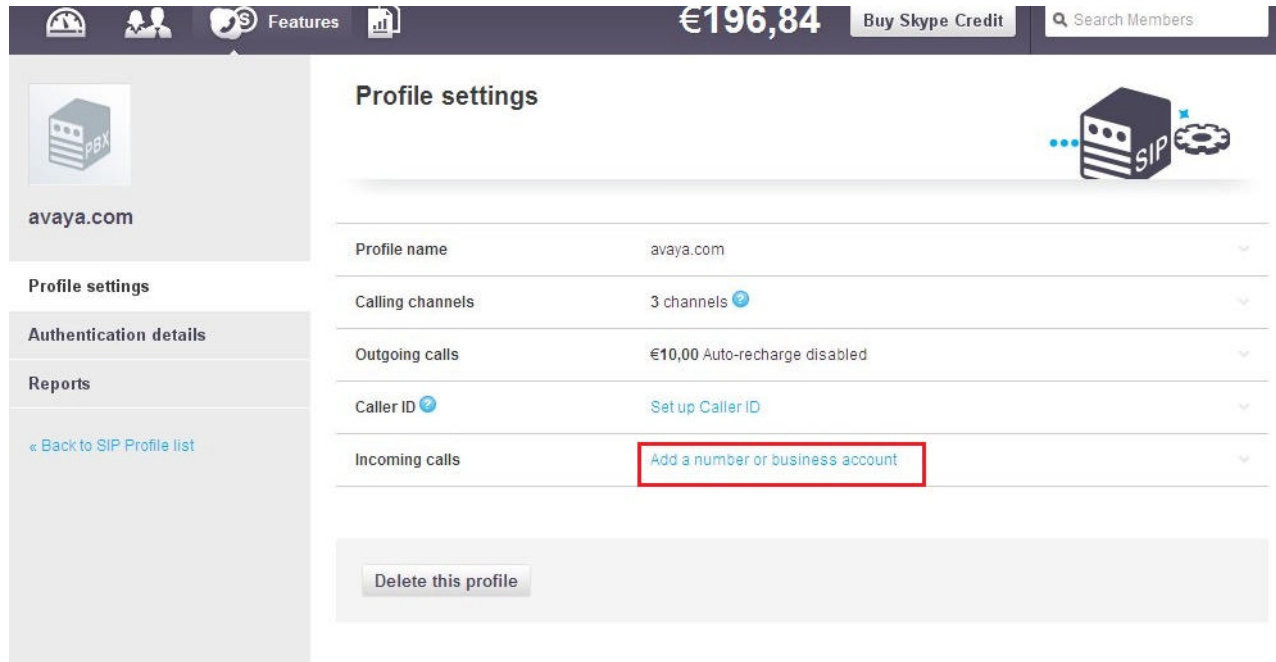
+12024706792

+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 Profile settings page for the user 'avaya.com'. The page has a dark header with icons for features and a balance of €196,84. The left sidebar contains navigation links: Profile settings, Authentication details, Reports, and a link back to the SIP Profile list. The main content area is titled 'Profile settings' and contains a list of settings:

Setting	Value
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, there 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**.

The screenshot shows the 'Profile settings' page for 'avaya.com'. The left sidebar contains navigation links: 'Profile settings', 'Authentication details', 'Reports', and a 'Back to SIP Profile list' link. The main content area is titled 'Profile settings' and includes a SIP icon. Below the title, there are several settings rows: '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 highlighted in light blue and contains the text: 'Add a number or business account'. Below this, a message states: '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.' There are two buttons: 'Add Online Number' and 'Add business account'. Under 'Add Online Number', there is a dropdown menu labeled 'Add a number' with the value '+12024703313' selected. Below the dropdown is a blue button labeled 'Allocate number' and a grey button labeled 'Buy a new number'. The 'Allocate number' button is highlighted with a red rectangle.

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile settings

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**.



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	▼
Incoming calls	 +12024703313	▼
	Add a number or business account	▼

Delete this profile

Click **Use and Online Number**.



avaya.com


Profile settings


Authentication details

Reports

[« Back to SIP Profile list](#)

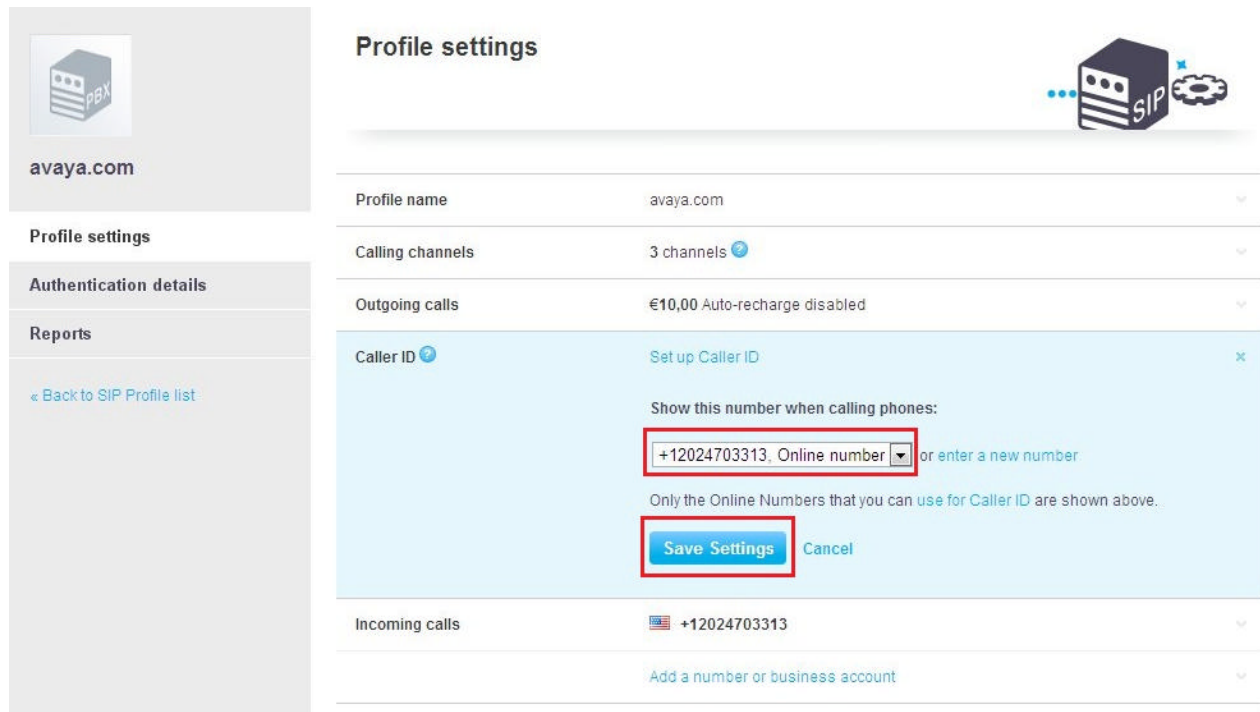
Profile settings



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

Delete this profile

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



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](#) 

Show this number when calling phones:

or [enter a new number](#)

Only the Online Numbers that you can use for Caller ID are shown above.

[Save Settings](#) [Cancel](#)

Incoming calls:  +12024703313

[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.

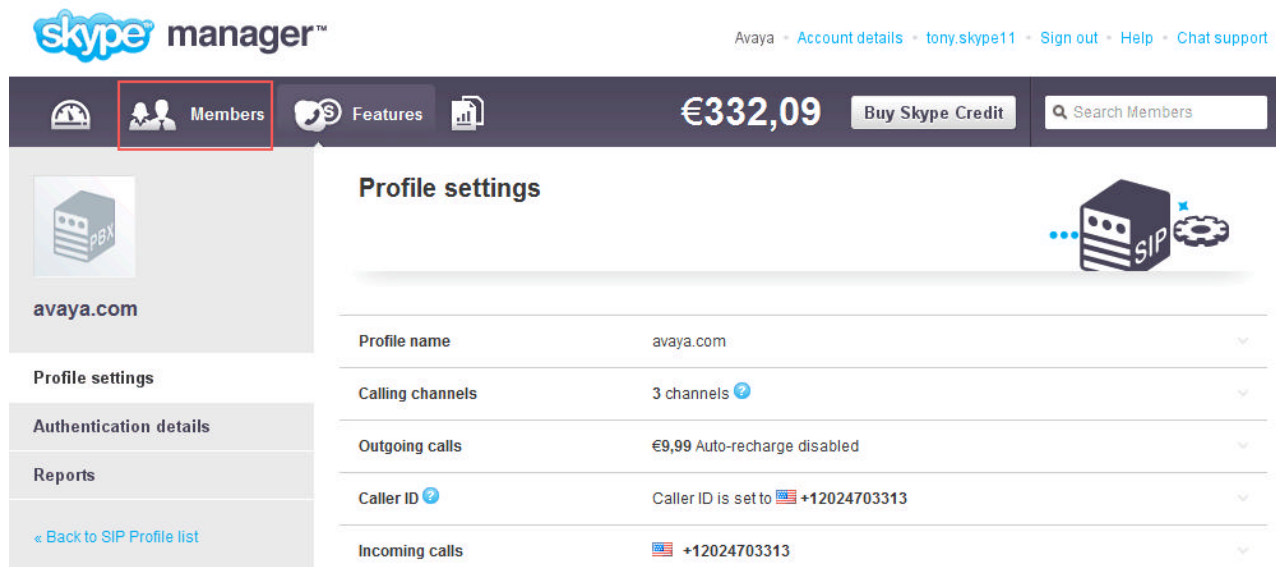
Profile settings

Profile name	avaya.com	
Calling channels	3 channels	
Outgoing calls	€10,00 Auto-recharge disabled	
Caller ID	Caller ID is set to +12024703313	
	Change Caller ID Disable Caller ID	
	Manage stored landline numbers	
Incoming calls	+12024703313	
	Add a number or business account	

[Delete this profile](#)

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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€332,09 Buy Skype Credit Search Members

Members Features

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile settings

Profile name avaya.com

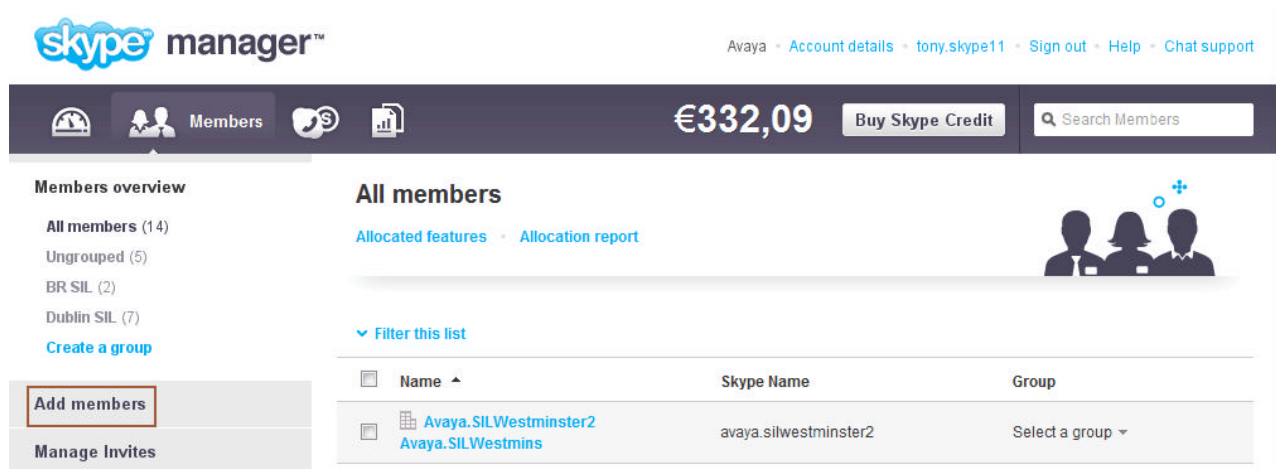
Calling channels 3 channels

Outgoing calls €9,99 Auto-recharge disabled

Caller ID Caller ID is set to +12024703313

Incoming calls +12024703313

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



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Members Features

Members overview

All members (14)

Ungrouped (5)

BR SIL (2)

Dublin SIL (7)

Create a group

Add members

Manage Invites

All members

Allocated features Allocation report

Filter this list

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

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

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€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**.

skype manager™

Avaya • Account details • tony.skype11 • Sign out • Help • Chat support

€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*

lbbob@avaya.com bob. avaya.com

First name bob

Last name smith

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 Ungrouped or [create a group](#)

Create accounts

6.8. Add Business User to SIP Profile

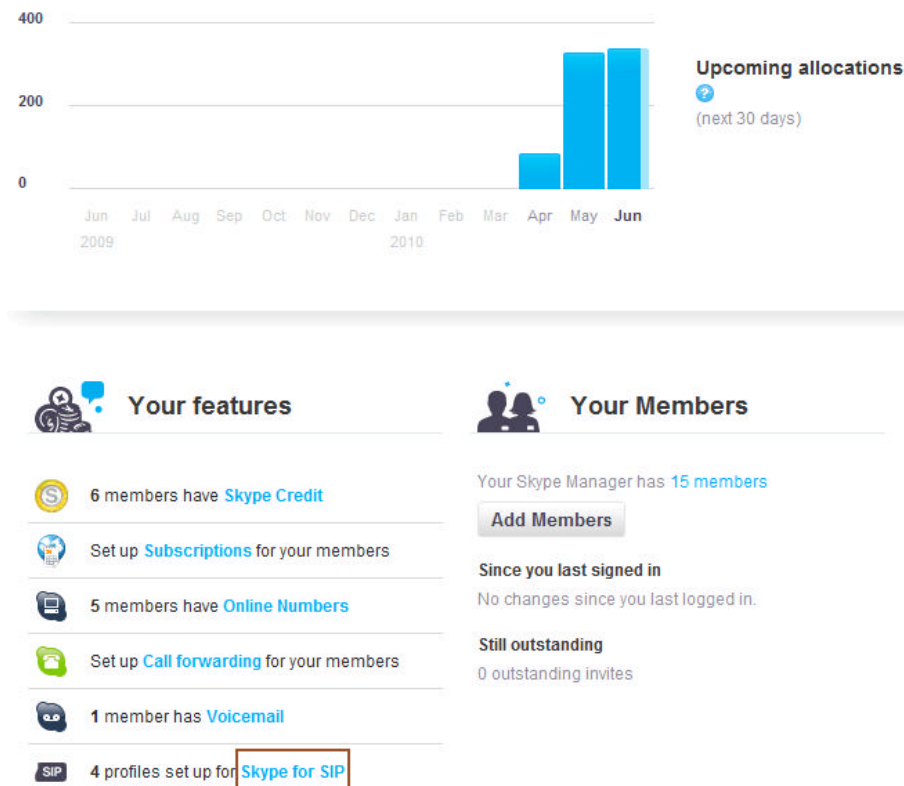
The **Create business accounts - Summary** page is displayed. Click **Dashboard**.

The screenshot shows the 'Create business accounts' summary page. The top navigation bar includes a 'Members' tab, a balance of €332,09, a 'Buy Skype Credit' button, and a search bar. The left sidebar has 'Members overview' and 'Add members' options. The main content area shows the progress: 1. Enter email addresses, 2. Enter details, 3. Summary (current). Below this, it states 'The following 1 accounts were created' and provides an email invitation link. A table lists the created account:


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


An 'Add to Skype contact list' button is located below the table.


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





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
	<div>Add a number or business account</div>

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 ?	▼
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 [×](#)

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

GB; Reviewed:
SPOC 10/1/2010

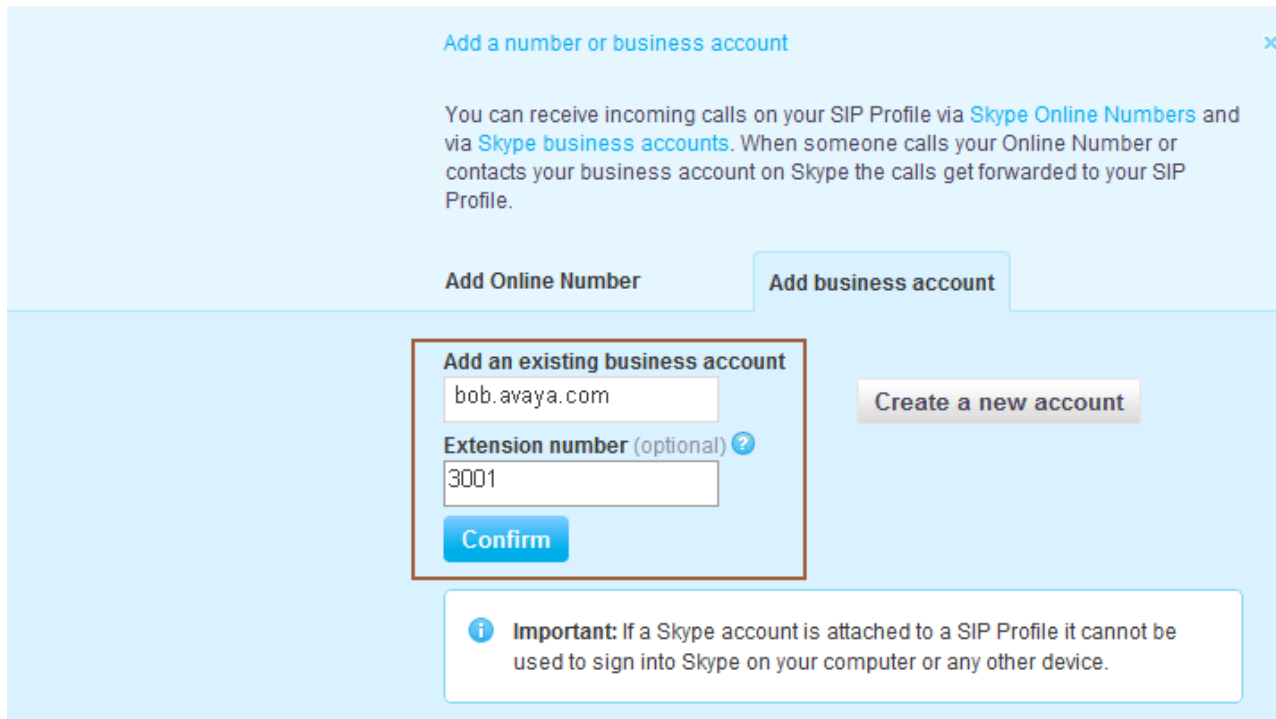
Solution & Interoperability Test Lab Application Notes
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66 of 89
Skype-AuraR6

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

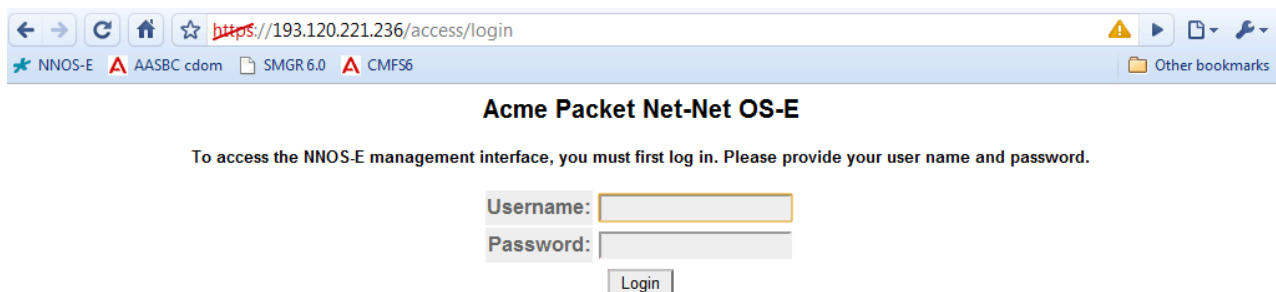
This section provides the procedures for configuring Avaya Aura™ Session Border Controller and includes the following items:

- 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.



The screenshot shows a web browser window with the address bar displaying `https://193.120.221.236/access/login`. The browser's bookmark bar contains entries for NNOS-E, AASBC cdom, SMGR 6.0, and CMFS6. The page title is "Acme Packet Net-Net OS-E". The main content area contains the text: "To access the NNOS-E management interface, you must first log in. Please provide your user name and password." Below this text are two input fields: "Username:" and "Password:". A "Login" button is positioned below the password field.

7.2. Administer SIP Domains

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

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

The screenshot shows the Acme Packet administration interface. The top navigation bar includes tabs for Home, Configuration (highlighted with a red box), Status, Call Logs, 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 various system metrics and status information in a table-like format.

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

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-po		
+ 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**.

Configuration: all

ConfigurationSetupView

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

server PBX1

sip-gateway Telco

dns-group group1

accounting

dns

settings

Configure vsp\enterprise\servers\sip-gateway PBXShow advanced

SetResetBackCopyDelete

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

general:

* namePBX

adminenabled (Resource is active)

domainavaya.com

failover-detectionping (Use OPTIONS to detect failures)

servers:

server-pool

Delete

GB; Reviewed:
SPOC 10/1/2010

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72 of 89
Skype-AuraR6

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**.

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 acmeApocket Configuration page. The left navigation pane is expanded to show the hierarchy: **vsp** → **session-config-pool** → **entry ToTelco** → **to-uri-specification**. The main content area displays "Configuration Loaded" and "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 configuration form for "Configure vspsession-config-poolentry ToTelcto-uri-specification". The form includes fields for user, host, port, display, transport, user-param, user-truncate-non-digits, and uri-parameter. The "host" field is highlighted with a red box, showing the "next-hop-domain" selected in the drop-down menu.

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**.

Configuration: all

Configuration Setup View

- cluster
 - box:aasbc
- vsp
 - default-session-config
 - tls
 - session-config-pool
 - dial-plan
 - enterprise
 - servers
 - sip-gateway PBX
 - sip-gateway Telco
 - vsp\session-config-pool\entry ToTelco
 - to-uri-specification
 - from-uri-specification**
 - request-uri-specification
 - p-asserted-identity-uri-specification
 - server-pool
 - dns settings

Configure vsp\session-config-pool\entry ToTelco\from-uri-specification Help Index

Set Reset Back Delete

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.)

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**.

Configuration: all

Configuration Setup View

- cluster
 - box:aasbc
- vsp
 - default-session-config
 - pre-session-config
 - session-config-pool
 - entry ToTelco
 - to-uri-specification
 - from-uri-specification
 - request-uri-specification**
 - p-asserted-identity-uri-specification
 - entry ToPBX
 - entry Discard
 - dial-plan registration-plan

Configure vsp\session-config-pool\entry ToTelco\request-uri-specification Help Index

Set Reset Back Delete

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.)

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**.

Configuration: all

Configuration Setup View

- cluster
 - box: aasbc
- vsp
 - default-session-config
 - pre-session-config
 - session-config-pool
 - entry ToTelco
 - to-uri-specification
 - from-uri-specification
 - request-uri-specification
 - p-asserted-identity-uri-specification**
 - entry ToPBX

Configure vsp\session-config-pool\entry ToTelco\p-asserted-identity-uri-specification

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

Set **Reset** **Back** **Delete**

user	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)
host	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.)
port	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.

Configuration: all

Configuration Setup View

- cluster
 - box:aasbc
- vsp
 - default-session-config
 - pre-session-config
 - session-config-pool
 - entry ToTelco
 - entry ToPBX
 - to-uri-specification**
 - from-uri-specification
 - request-uri-specification
 - p-asserted-identity-uri-specification

Configure vsp\session-config-pool\entry ToPBX\to-uri-specification Help Index

Set Reset Back Delete

user	enter <input type="text" value="to-uri"/> or select from <input type="text" value="to-uri"/> (Net-Net OS-E uses the value from the incoming TO URI.)
host	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.)
port	enter <input type="text" value="to-uri"/> or select from <input type="text" value="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**.

Configuration: all

Configuration Setup View

- cluster
 - box:aasbc
- vsp
 - default-session-config
 - pre-session-config
 - session-config-pool
 - entry ToTelco
 - entry ToPBX
 - to-uri-specification
 - from-uri-specification**
 - request-uri-specification
 - p-asserted-identity-uri-specification

Configure vsp\session-config-pool\entry ToPBX\from-uri-specification Help Index

Set Reset Back Delete

user	enter <input type="text" value="request-uri"/> or select from <input type="text" value="request-uri"/> (Net-Net OS-E uses the value from the incoming REQUEST URI.)
host	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.)
port	enter <input type="text" value="from-uri"/> or select from <input type="text" value="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 structure under 'vsp' with 'session-config-pool' expanded, and 'entry ToPBX' selected. The 'request-uri-specification' option is highlighted. The main pane shows the configuration for 'request-uri-specification'. It has buttons for 'Set', 'Reset', 'Back', and 'Delete'. The configuration table has three rows: 'user', 'host', and 'port'. The 'host' row has a dropdown menu 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 structure under 'vsp' with 'session-config-pool' expanded, and 'entry ToPBX' selected. The 'p-asserted-identity-uri-specification' option is highlighted. The main pane shows the configuration for 'p-asserted-identity-uri-specification'. It has buttons for 'Set', 'Reset', 'Back', and 'Delete'. The configuration table has three rows: 'user', 'host', and 'port'. The 'host' row has a dropdown menu 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, the 'Configuration: all' tree is expanded to 'vsp' > 'pre-session-config' > 'sip-header-settings'. 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' section shows 'enabled' with a dropdown arrow and the text '(Resource is active)'. The 'rule' section has an 'Add rule' button. 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 'Create rule' dialog box. On the left, the 'Configuration: all' tree is expanded to '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' page is displayed. It contains the instruction 'Please provide some basic information for rule. Then press "Create".' Below this, there is a text input field labeled '* name' with 'P-Site' entered. At the bottom, there are 'Create', 'Reset', and 'Cancel' buttons.

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

Set	Reset	Back	Copy	Delete
-----	-------	------	------	--------

* name	P-Site
description	
condition	Configure
action	Configure

Set	Reset	Back	Copy
-----	-------	------	------

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

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**.

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\condition - Step 1 of 1: Edit condition

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

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

* condition-type	match-header	(Sets the name of the header to match on)
* name	enter P-Site or select from	<Not configured>

Create	Reset	Cancel
--------	-------	--------

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

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

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 'Update and save configuration' is highlighted. The main pane shows the configuration for 'rule P-Site'. The configuration details are as follows:

Configure vsp\pre-session-config\sip-header-settings\rule P-Site	
* name	P-Site
description	
condition	<p>* condition-type: match-header (Sets the name of the header to match on)</p> <p>* name: enter P-Site or select from P-Site</p>
action	<p>* action-type: strip-header (The Net-Net OS-E removes the SIP header from the packet)</p>

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 titled 'Configuration Updated and Saved' is displayed, stating 'The running configuration has been updated and saved.' The main pane shows the configuration for 'rule P-Site'.

Configure vsp\pre-session-config\sip-header-settings\rule P-Site	
* name	P-Site
description	
condition	<p>* condition-type: match-header (Sets the name of the header to match on)</p> <p>* name: enter P-Site or select from P-Site</p>
action	<p>* action-type: strip-header (The Net-Net OS-E removes the SIP header from the packet)</p>

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 displays 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 shows 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). At the bottom, the 'master-services' section lists 'accounting, database'.

box-identifier	013e-d911-d96b-767d
box-status	
IPAddress	LocalBox (193.120.221.236)
State	Connected
build-version	3.6.0
build-number	46303M-dev
master-services	
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** seconds Refresh

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

Search Type: All Sessions View All Sessions Search

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.

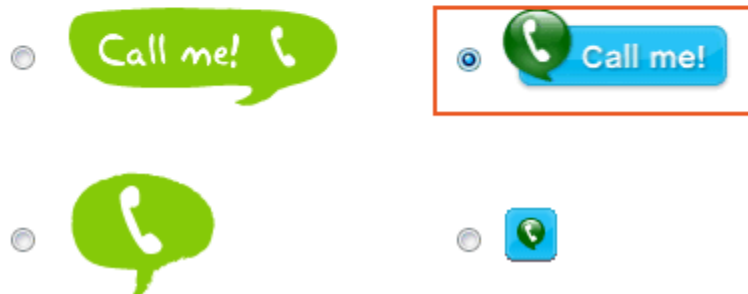
Get a Skype button

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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