



Application Notes for Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 and Avaya Aura® Session Border Controller 6.0 with Skype Connect R2.2 using TLS as the Transport Protocol – 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 using TLS as the transport protocol.

Skype Connect allows Skype's 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	7
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.	Outbound to Skype Client	12
4.	Avaya Aura® Communication Manager	13
4.1.	Administer System Parameters	13
4.2.	Administer Dial Plan.....	14
4.3.	Administer IP Network Parameters.....	15
4.4.	Administer Signaling Group	17
4.5.	Administer Trunk Group.....	18
4.6.	Administer Automatic Route Selection.....	19
4.7.	Administer Route Pattern	20
4.8.	Administer Public Unknown Numbering.....	20
4.9.	Administer Incoming Call Handling Treatment.....	21
5.	Configuring Avaya Aura® Session Manager	22
5.1.	Log in to Avaya Aura® Session Manager	22
5.2.	Administer SIP Domain	24
5.3.	Administer Locations	25
5.4.	Administer SIP Entities	26
5.5.	Administer Entity Links	28
5.6.	Administer Time Ranges.....	29
5.7.	Administer Routing Policies	30
5.8.	Administer Dial Patterns	32
6.	Configure Skype Manager	35
6.1.	Log in to Skype Manager	35
6.2.	Create SIP Profile.....	36
6.3.	Administer Maximum Simultaneous Calls	37
6.4.	Administer Online Numbers	40
6.5.	Administer Caller ID.....	44
6.6.	Create a Business User	46
6.7.	Add Business User to SIP Profile.....	48
7.	Configure Avaya Aura® Session Border Controller	51
7.1.	Log in to Avaya Aura® Session Border Controller using the GUI.....	51
7.2.	Administer SIP Domains.....	52

7.3.	Administer TLS Certificate	56
7.4.	Administer TLS on the outside interface	58
7.5.	Administer Authentication	60
7.6.	Administer Outbound SIP Header Manipulation Rules	61
7.7.	Administer Inbound SIP Header Manipulation Rules.....	64
7.8.	Administer SIP Header Rules for Topology Hiding	66
7.9.	Save the Configuration	70
8.	General Test Approach and Test Results.....	71
9.	Verification Steps.....	71
9.1.	Troubleshooting Tools	72
10.	Conclusion	72
11.	References.....	73
	Appendix A – Skype Button	74
	Appendix B – Avaya Aura® SBC Configuration File	76

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 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.

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 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 are 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 currently supported by Skype Connect R2.2
- 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 and SRV records are supported for Skype Connect R2.2
- The SIP REFER request is not supported for call redirection/transfer.
- SIP 3xx Redirect Responses are not supported.
- SRTP is supported in Skype Connect R2.2 and was tested successfully on the SIP trunk towards the SBC public interface.
- 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 (Annex A and B) 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 the format <country code><national number>.
- T.38 fax is not supported.
- RTCP and RTCP XR are not handled by Skype Connect.
- 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.
- 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.
- No RTP was received from Skype when calling an international number (448458444777). This issue is currently being investigated by Skype.

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 TLS to TCP for inbound and vice-versa for outbound calls.

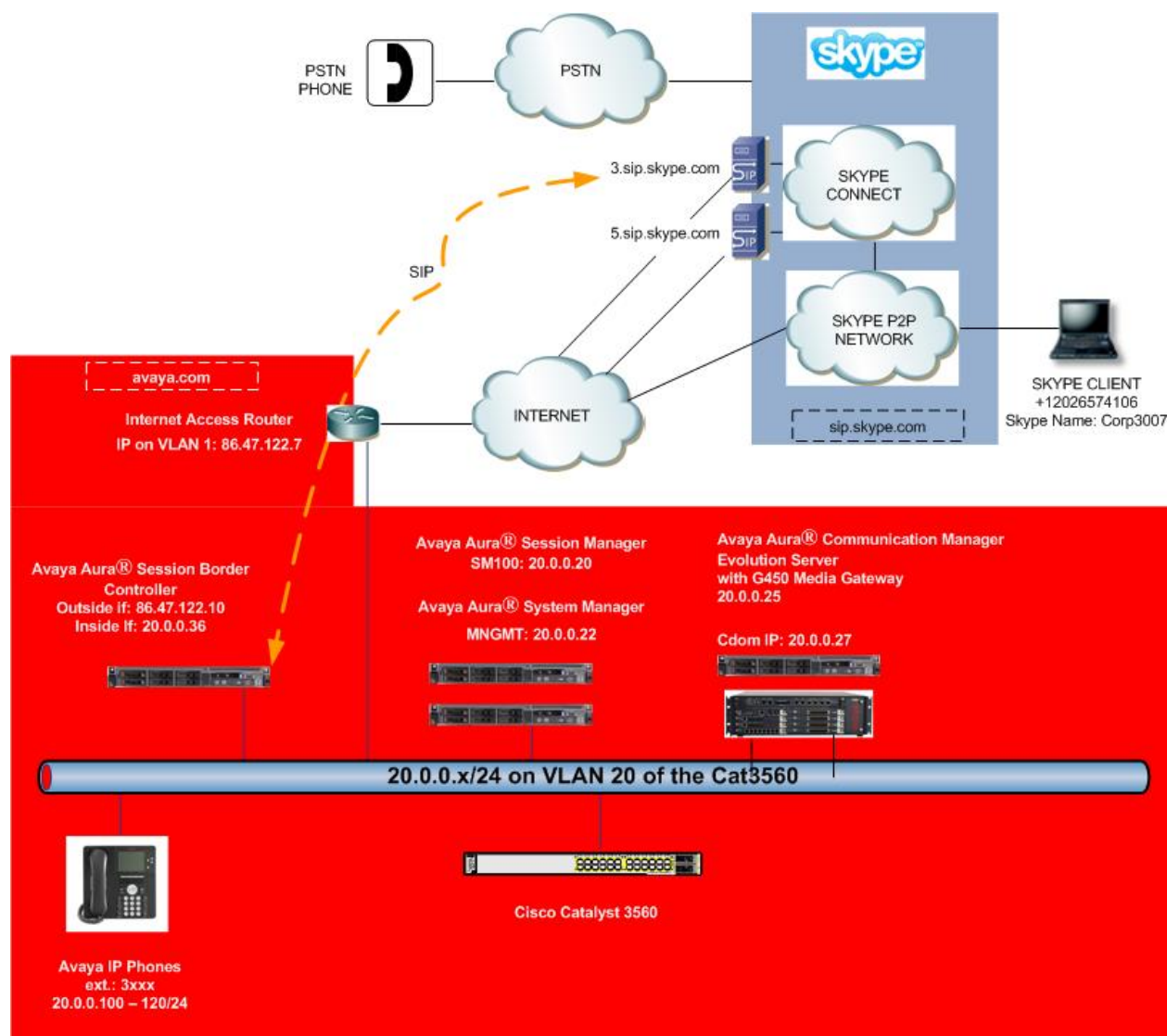


Figure 1: Avaya Interoperability Test Lab Reference Configuration

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 – R6.1 Build R6.1.0.610012
Avaya S8800 Server	Avaya Aura [®] System Manager - R6.1 Build R6.1.0.610012
Avaya S8800 Server	Avaya Aura [®] Session Border Controller – R6.0.0.1.5 Build E36m3p4
Avaya S8800 Server	Avaya Aura [®] Communication Manager – Evolution Server - R016x.00.1.510.1
Avaya G430 Media Gateway	R30.10.4
Avaya 9650 H.323 Phone	R3.1.1
Avaya 9650 SIP Phone	R2.6.2.21
Test PC1	Microsoft Windows Vista with Avaya one-X [®] Communicator (H.323) R6.0.0.20
Test PC2	Skype Client v5.3.0.111
Skype Connect	R2.2 Build 3.6.0m3p4

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 Session Border Controller, Session Manager and 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 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 Session Border Controller.
4. The call is delivered to the Session Border Controller using SIP over TLS.
5. The Session Border Controller performs SIP header modifications, SDP modifications, transport protocol conversion (TLS to TCP) and routes the call to Session Manager.
6. 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 Communication Manager SIP Entity.
7. Communication Manager routes the call to a phone.

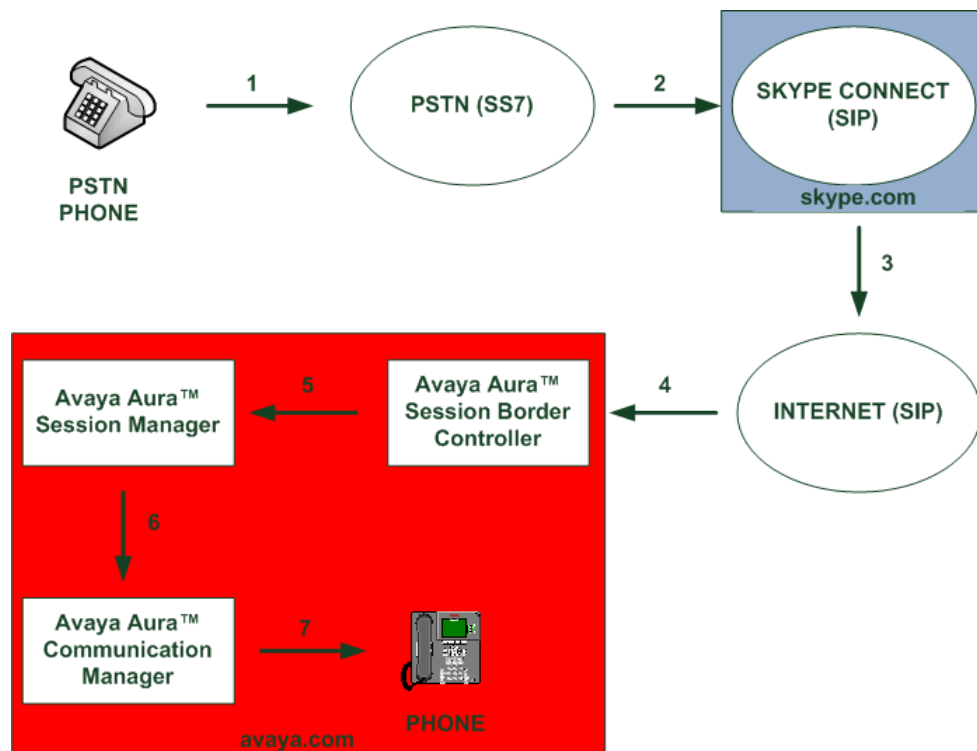


Figure 2 - Inbound PSTN to Skype Connect Call

Note: A single 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 Communication Manager on the enterprise network and terminates at the PSTN phone.

1. Avaya phone originates a call to a PSTN number.
2. Communication Manager converts the calling and called numbers to an E.164 format before it routes the call to Session Manager using SIP over TCP.
3. Based on the called number Session Manager sends the call to Avaya Aura® Session Border Controller.
4. 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 SRV record. The call is sent using SIP over TLS 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 E.164 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.

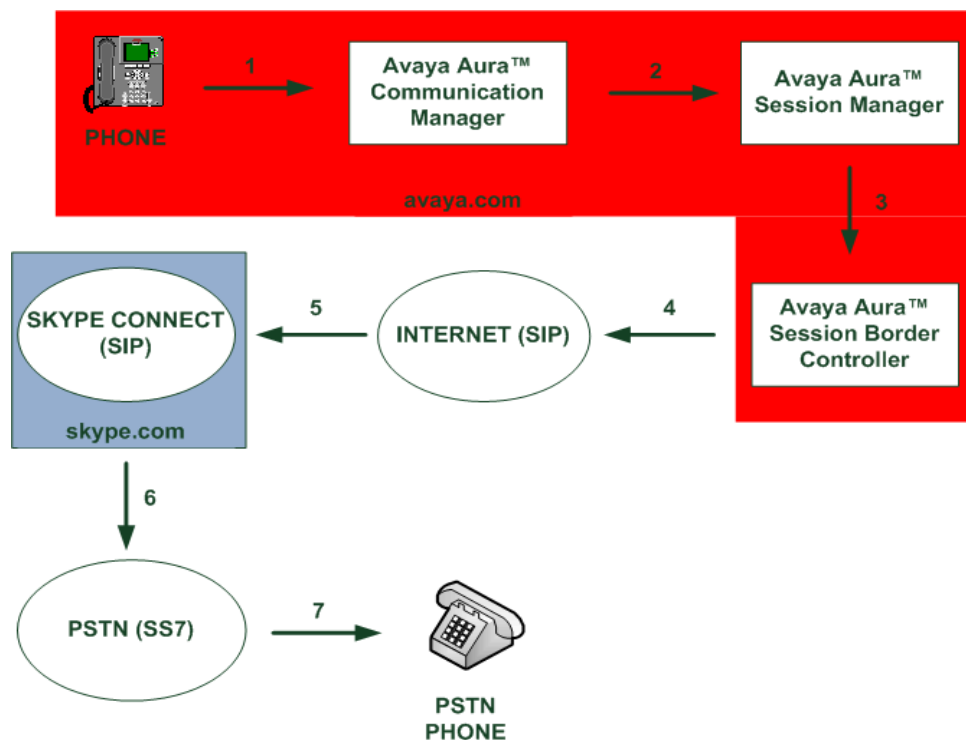


Figure 3: Outbound Skype Connect to PSTN call with CLIP

Note: Session Border Controller caches DNS SRV 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 TLS.
5. The Avaya Aura® Session Border Controller performs SIP header modifications, SDP modifications, transport protocol conversion (TLS 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.

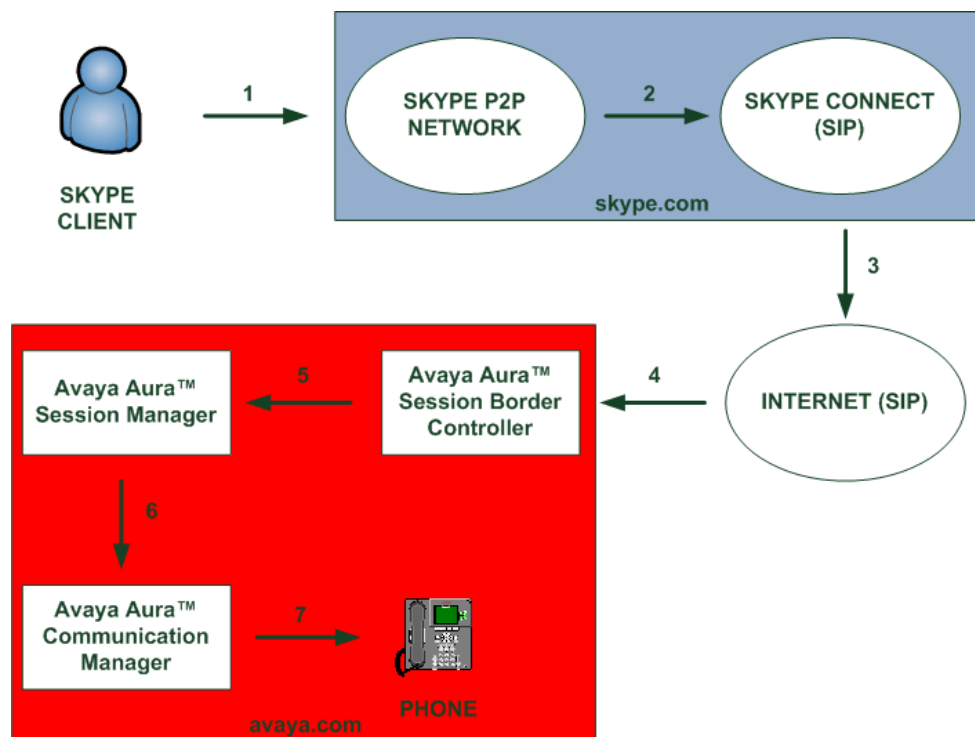


Figure 4: Inbound Skype P2P to Skype Connect call

3.4. Outbound to Skype Client

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 E.164 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 SRV record. The call is sent using SIP over TLS 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.

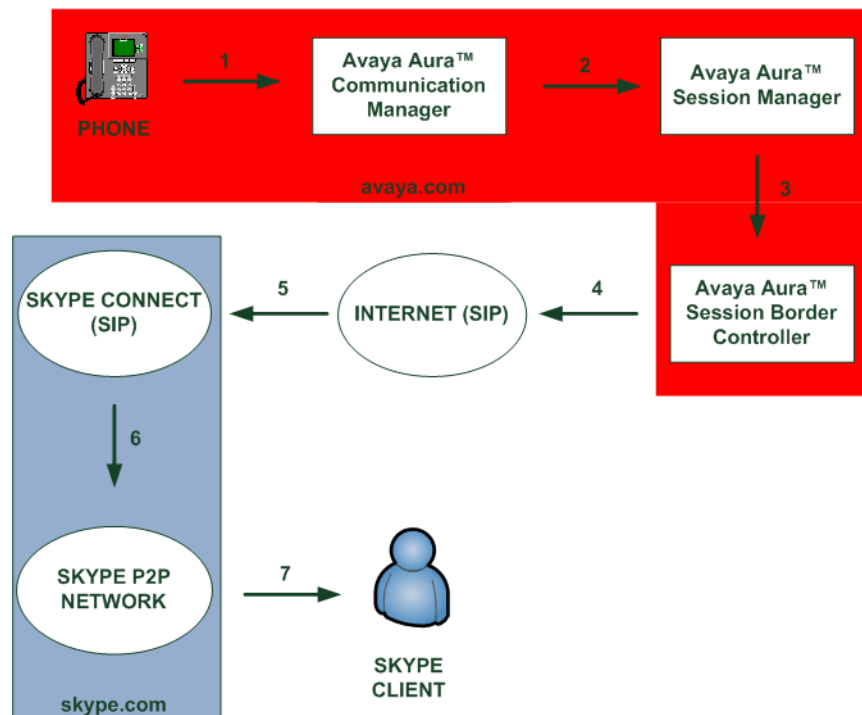


Figure 5: Outbound Skype Connect to Skype P2P Client call

4. 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.

4.1. 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		

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

4.3. 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, G430 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		
Codec Set: 1	Intra-region IP-IP Direct Audio: yes	
UDP Port Min: 2048	Inter-region IP-IP Direct Audio: yes	
UDP Port Max: 3329	IP Audio Hairpinning? N	
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46	RTCP Reporting Enabled? Y	
Audio PHB Value: 46	RTCP MONITOR SERVER PARAMETERS	
Video PHB Value: 26	Use Default Server Parameters? Y	
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		
Codec Set: 2	Intra-region IP-IP Direct Audio: yes	
UDP Port Min: 2048	Inter-region IP-IP Direct Audio: yes	
UDP Port Max: 3329	IP Audio Hairpinning? N	
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46	RTCP Reporting Enabled? Y	
Audio PHB Value: 46	RTCP MONITOR SERVER PARAMETERS	
Video PHB Value: 26	Use Default Server Parameters? Y	
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 for connections to destination region 1.

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

4.4. 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 SM100 ip address
- **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
IMS Enabled? n	Transport Method: tcp
Q-SIP? n	SIP Enabled LSP? n
IP Video? n	Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y	Peer Server: SM
Near-end Node Name: procr	Far-end Node Name: asmr61
Near-end Listen Port: 5060	Far-end Listen Port: 5060
	Far-end Network Region: 2
Far-end Domain:	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y	IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n
	Alternate Route Timer(sec): 6

4.5. 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: 100	
		Number of Members: 20	

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

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

add trunk-group 100		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none		
		Maintenance Tests? y	
Numbering Format: public			
		UUI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
Show ANSWERED BY on Display? y			

4.6. 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.

change ars analysis 0							Page	1 of	2
ARS DIGIT ANALYSIS TABLE									
Location: all							Percent Full:	0	
	Dialed	Total		Route	Call	Node	ANI		
	String	Min	Max	Pattern	Type	Num	Reqd		
00		10	18	100	pubu		n		

4.7. Administer Route Pattern

Use the **change route-pattern** command to define the trunk group administered in **Section 4.5**.

change route-pattern 100										Page 1 of 3	
Pattern Number: 68 Pattern Name:toasmr61											
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

4.8. 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 E.164 verified landline numbers that are associated with the SIP Profile in Skype Manager minus the “+” as this will be added automatically by Communication Manager for calls that use a public trunk. In **Section 6.5** 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						
Ext	Ext	Trk	CPN	Total		
Len	Code	Grp(s)	Prefix	CPN		
				Len		
5	3001	100	12024703313	11	Total Administered: 2	
					Maximum Entries: 9999	

4.9. 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 6.4** the Online Number is set to +12024703313 in Skype Manager but it should be noted that in the request-uri that Skype sends, the “+” is omitted.

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			

5. Configuring Avaya Aura[®] Session Manager

This section provides the procedures for configuring Session Manager. For further information on Session Manager, please consult with references [1]-[3]. The procedures include the following areas:

- Log in to Avaya Aura[®] Session Manager
- Administer SIP domain
- Administer Locations
- Administer SIP Entities
- Administer Entity Links
- Administer Time Ranges
- Administer Routing Policies
- Administer Dial Patterns

The administration of Avaya endpoints and Communication Manager is not covered in these Application Notes. Please consult reference [4].

5.1. Log in to Avaya Aura[®] Session Manager

Access the System Manager using a Web Browser and entering ***http://<ip-address>/SMGR***, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials.

AVAYA Avaya Aura™ System Manager 6.1

Home / Log On

Log On

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

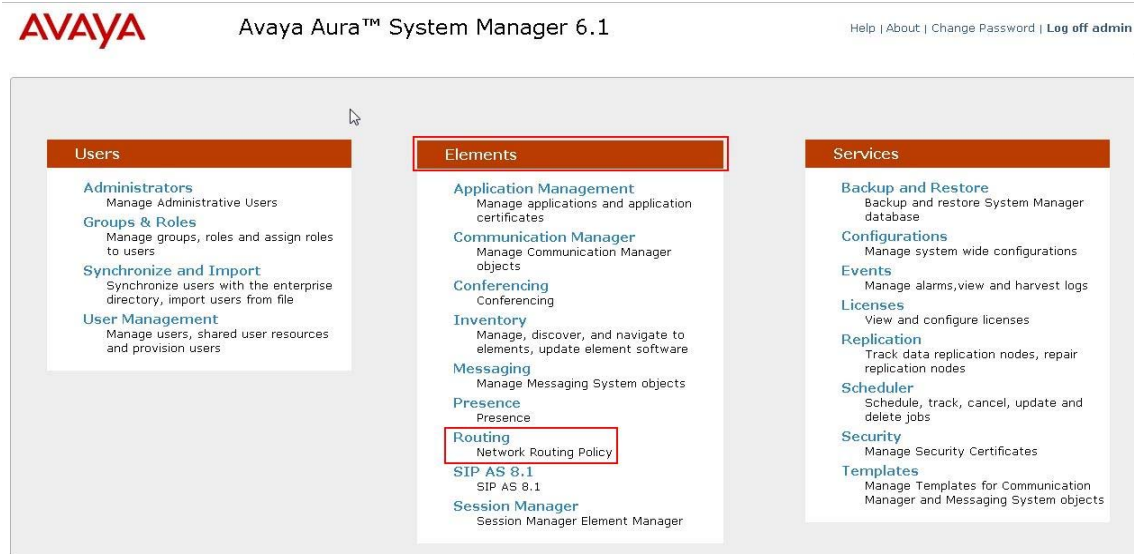
Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

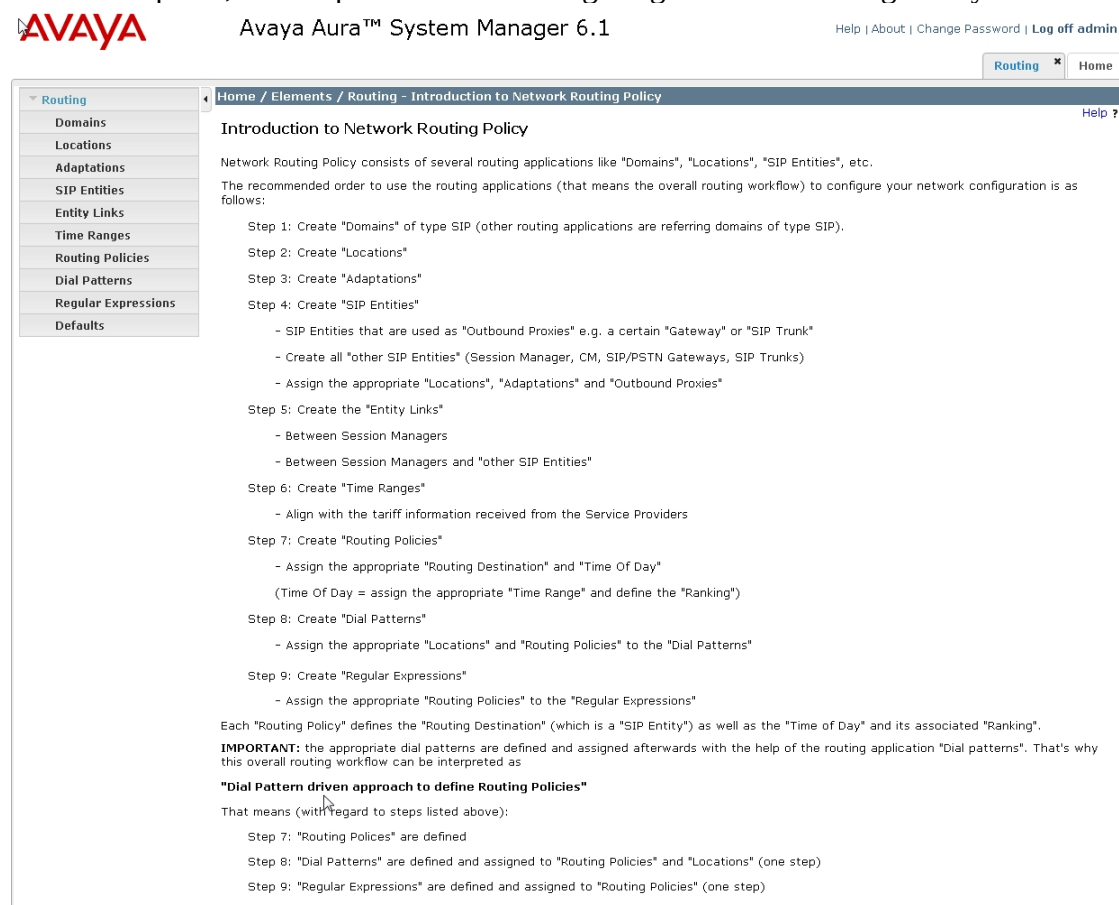
Password:

[Change Password](#)

In the next screen under **Elements** column select **Routing**.



In the main panel, a short procedure for configuring Network Routing Policy is shown.



5.2. Administer SIP Domain

Add the SIP domain, for which the communications infrastructure will be authoritative, by selecting **Domains** on the left panel menu and clicking the **New** button (not shown) to create a new SIP domain entry. Complete the following options:

- **Name** The authoritative domain name (e.g., **avaya.com**)
- **Type** **SIP**
- **Notes** Description for the domain (optional)

Click **Commit** to save changes.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar has a 'Routing' section with 'Domains' selected. The main area is titled 'Domain Management' and shows a table with one row for 'avaya.com' with type 'sip'. The 'Commit' button is highlighted.

Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Domains - Domain Management

Domain Management

1 Item Refresh Filter: Enable

Name	Type	Default	Notes
* avaya.com	sip	☐	

* Input Required

Commit Cancel

5.3. Administer Locations

Session Manager uses the origination location to determine which dial patterns to look at when routing the call if there are dial patterns administered for specific locations. Locations are also used to limit the number of calls coming out of or going to a physical location. This is useful for those locations where the network bandwidth is limited. To add a Location, select **Locations** on the left panel menu and then click the **New** button (not shown). Enter the following for each **Location**:

Under **General**:

- **Name** An informative name (e.g. enterprise)

Under **Location Pattern**:

- **IP Address Pattern** An IP address pattern for this location

Select **Add** to add more locations. Click **Commit** to save changes.

AVAYA Avaya Aura™ System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Locations - Location Details

Location Details

Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. See Session Manager -> Session Manager Administration -> Global Setting

General

* Name: enterprise

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Per-Call Bandwidth Parameters

* Default Audio Bandwidth: 80 Kbit/sec

Location Pattern

Add Remove

1 Item Refresh Filter: Enable

IP Address Pattern	Notes
* 20.0.0.*	

Select : All, None

* Input Required

Commit Cancel

5.4. Administer SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by a SIP Trunk. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). Enter the following for each SIP Entity:

Under **General**:

- **Name** An informative name (e.g., **r61sm**)
- **FQDN or IP Address** IP address of the signaling interface on the Session Manager, Communication Manager or SBC.
- **Type** **Session Manager** for Session Manager, **CM** for Communication Manager and **SIP Trunk** for SBC.
- **Location** **enterprise**
- **Time Zone** Time zone for this location

The following screen shows the SIP Entity for Session Manager.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a menu with 'SIP Entities' highlighted. The main area displays the 'SIP Entity Details' form for a 'Session Manager' entity. The form fields are: Name (r61sm), FQDN or IP Address (20.0.0.20), Type (Session Manager), Notes (Session Manager R6.1), Location (Enterprise), Outbound Proxy (empty), Time Zone (Etc/GMT), and Credential name (empty). The 'Commit' button is highlighted in the top right corner.

Under **Port**, click **Add**, and then edit the fields in the resulting new row.

- **Port** Port number on which the system listens for SIP requests
- **Protocol** Transport protocol to be used to send SIP requests

The following screen shows the Port definitions for the Session Manager SIP Entity. Click **Commit** to save changes.

The screenshot shows the 'Port' definitions table for the Session Manager SIP Entity. The table has columns: Port, Protocol, Default Domain, and Notes. Two rows are shown: Port 5060 with Protocol TCP and Default Domain avaya.com, and Port 5061 with Protocol TLS and Default Domain avaya.com. The 'Add' button is highlighted in the top left corner. The 'Commit' button is highlighted in the bottom right corner.

Port	Protocol	Default Domain	Notes
5060	TCP	avaya.com	
5061	TLS	avaya.com	

The following screen shows the SIP Entity for Communication Manager.

AVAYA Avaya Aura™ System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

General

* Name: CMES61

* FQDN or IP Address: 20.0.0.25

Type: CM

Notes:

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

Commit Cancel

The following screen shows the SIP Entity for SBC.

AVAYA Avaya Aura™ System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

General

* Name: AASBCr6

* FQDN or IP Address: 20.0.0.36

Type: SIP Trunk

Notes: Avaya Aura SBC R6

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: egress

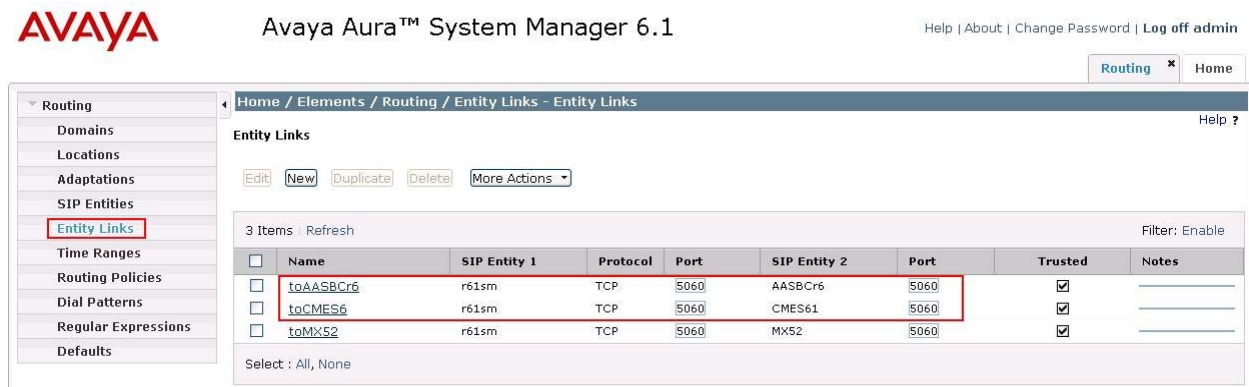
Commit Cancel

5.5. Administer Entity Links

A SIP trunk between a Session Manager and a telephony system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- **Name** An informative name
- **SIP Entity 1** Select **r61sm**
- **Port** Port number to which the other system sends its SIP requests
- **SIP Entity 2** The other SIP Entity for this link, created in **Section 5.4**
- **Port** Port number to which the other system expects to receive SIP requests
- **Trusted** Whether to trust the other system
- **Protocol** Transport protocol to be used to send SIP requests

Click **Commit** to save changes. The following screen shows the Entity Links used in the sample network.



Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Entity Links - Entity Links

Entity Links

Edit New Duplicate Delete More Actions

3 Items Refresh Filter: Enable

☐	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
☐	toAASBCr6	r61sm	TCP	5060	AASBCr6	5060	☒	
☐	toCMES6	r61sm	TCP	5060	CMES61	5060	☒	
☐	toMX52	r61sm	TCP	5060	MX52	5060	☒	

Select : All, None

5.6. Administer Time Ranges

Before adding routing policies (see next step), time ranges must be defined during which the policies will be active. To add this time range, select **Time Ranges** from the left panel menu and then click **New** on the right. Fill in the following fields.

- **Name** An informative name (e.g. always)
- **Mo through Su** Check the box under each day of the week for inclusion
- **Start Time** Enter start time (e.g. **00:00** for start of day)
- **End Time** Enter end time (e.g. **23:59** for end of day)

In Session Manager, a default policy (**24/7**) is available that would allow routing to occur at anytime. This time range was used in the sample network.

Routing x Home

Home / Elements / Routing / Time Ranges - Time Ranges Help ?

Time Ranges

Edit New Duplicate Delete More Actions

1 Item Refresh Filter: Enable

	Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
☐	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	

Select : All, None

5.7. Administer Routing Policies

Create routing policies to direct how calls will be routed to a system. Two routing policies must be added, one for Communication Manager (H.323 and Digital phones) and one for calls to Skype Connect numbers. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- Enter an informative **Name**

Under **SIP Entity as Destination**:

- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

Under **Time of Day**:

- Click **Add**, and then select the time range configured in the **Section 5.6**.

Click **Commit** to save changes. The following screen shows the **Routing Policy Details** for calls to Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left sidebar shows the navigation menu with 'Routing Policies' highlighted. The main content area is titled 'Routing Policy Details' and contains the following sections:

- General**: Includes a text field for 'Name' with the value 'toCMES61', a 'Disabled' checkbox, and a 'Notes' text area. 'Commit' and 'Cancel' buttons are in the top right.
- SIP Entity as Destination**: Features a 'Select' button.
- Table**: A table with columns 'Name', 'FQDN or IP Address', 'Type', and 'Notes'. It contains one entry: 'CMES61' with FQDN '20.0.0.25' and Type 'CM'.
- Time of Day**: Includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons.
- Time Range Table**: A table with columns 'Ranking', 'Name', 'Mon', 'Tue', 'Wed', 'Thu', 'Fri', 'Sat', 'Sun', 'Start Time', 'End Time', and 'Notes'. It shows one item with Ranking '0', Name '24/7', and Start/End times '00:00' to '23:59'. The 'Notes' column contains 'Time Range 24/7'.

At the bottom of the Time of Day section, there is a 'Select : All, None' option.

The following is screen shows the **Routing Policy Details** for calls to Skype Connect numbers.



Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

Help ?

Commit Cancel

General

* Name: toSkypeTLS

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
AASBCr6	20.0.0.36	SIP Trunk	Avaya Aura SBC R6

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

☐	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
☒	0	24/7	☒	☒	☒	☒	☒	☒	☒	00:00	23:59	Time Range 24/7

Select : All, None

5.8. Administer Dial Patterns

A dial pattern must be defined that will direct calls to the appropriate telephony system. In the sample network, calls to 4-digit extensions on Communication Manager will begin with **300**. To configure the extension Dial Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- **Pattern** Dialed number or prefix
- **Min** Minimum length of dialed number
- **Max** Maximum length of dialed number
- **Notes** Comment on purpose of dial pattern
- **SIP Domain** Select the SIP Domain specified in **Section 5.2**

Under **Originating Locations and Routing Policy** select **Add** (not shown). Select the following entries:

- **Originating Location Name** Select the location created in **Section 5.3**
- **Routing Policies** Select the policy **toCMES61** created in **Section 5.7**.

Select **Commit** button to save.

AVAYA Avaya Aura™ System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Help ?

Commit Cancel

General

* Pattern: 300x

* 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	Routing Policy Notes
☐	Enterprise	Private LAN	toCMES61	0	☐	CMES61	

Select : All, None

Denied Originating Locations

Add Remove

0 Items Refresh Filter: Enable

☐	Originating Location	Notes
--------------------------	----------------------	-------

Inbound calls from Skype Connect will begin with **1202**. To configure the Dial Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- **Pattern** Dialed number or prefix
- **Min** Minimum length of dialed number
- **Max** Maximum length of dialed number
- **Notes** Comment on purpose of dial pattern
- **SIP Domain** Select the SIP Domain specified in **Section 5.2**

Under **Originating Locations and Routing Policy** select **Add** (not shown). Select the following entries:

- **Originating Location Name** Select the location created in **Section 5.3**
- **Routing Policies** Select the policy **toCMES61** created in **Section 5.7**

Select **Commit** button to save.



Avaya Aura™ System Manager 6.1

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

[Routing](#) [Home](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Dial Patterns - Dial Pattern Details

Dial Pattern Details

Commit

Cancel

Help ?

General

* Pattern:

1202x

* Min:

5

* Max:

12

Emergency Call:

☐

SIP Domain:

avaya.com

Notes:

Originating Locations and Routing Policies

Add

Remove

1 Item

Refresh

Filter: Enable

☐	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	Enterprise	Private LAN	toCMES61	0	☐	CMES61	

Select : All, None

Denied Originating Locations

Add

Remove

0 Items

Refresh

Filter: Enable

☐	Originating Location	Notes
--------------------------	----------------------	-------

Outbound calls to Skype Connect numbers will begin with **00**. To configure the Dial Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- **Pattern** Dialed number or prefix
- **Min** Minimum length of dialed number
- **Max** Maximum length of dialed number
- **Notes** Comment on purpose of dial pattern
- **SIP Domain** Select the SIP Domain specified in **Section 5.2**

Under **Originating Locations and Routing Policy** select **Add** (not shown). Select the following entries:

- **Originating Location Name** Select the location created in **Section 5.3**
- **Routing Policies** Select the policy **toSkypeTLS** created in **Section 5.7**

Select **Commit** button to save.



Avaya Aura™ System Manager 6.1

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

[Routing](#) * [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#) - Dial Pattern Details

Help ?
Commit
Cancel

Routing

[Domains](#)
[Locations](#)
[Adaptations](#)
[SIP Entities](#)
[Entity Links](#)
[Time Ranges](#)
[Routing Policies](#)
Dial Patterns
[Regular Expressions](#)
[Defaults](#)

Dial Pattern Details

General

* Pattern: 00x

* Min: 5

* Max: 14

Emergency Call: ☐

SIP Domain: avaya.com

Notes:

Originating Locations and Routing Policies

Add Remove
1 Item | Refresh
Filter: Enable

	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
☐	Enterprise	Private LAN	toSkypeTLS	0	☐	AASBCr6	

Select : All, None

Denied Originating Locations

Add Remove
0 Items | Refresh
Filter: Enable

	Originating Location	Notes
☐		

TPN:Reviewed
SPOC 08/01/2011

Solution & Interoperability Test Lab Application Notes
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34 of 88
Skype-Avaya-TLS

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 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.

Note – When Skype Connect R2.2 becomes Generally Available (GA), there will be an option (tick box) to select TLS/SRTP in Skype Manager. However, during interoperability testing with the Skype Connect R2.2, there was no Skype Manager provisioning required for TLS/SRTP.

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**.

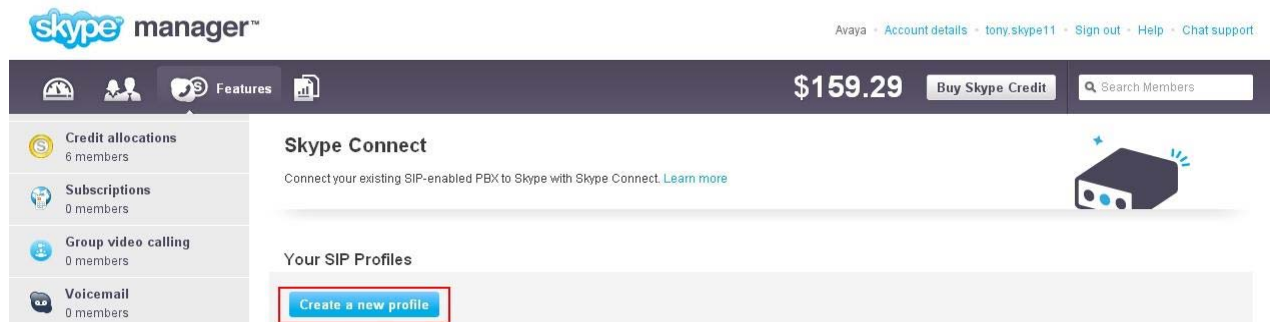
The screenshot shows the Skype Manager Dashboard. The top navigation bar includes 'Dashboard', 'Features' (highlighted with a red box), and a toolbar with the current balance of €226,94, a 'Buy Skype Credit' button, and a search bar. The main content area is divided into 'Reports' and 'Your account'. The 'Reports' section shows a bar chart for 'Allocations' from June 2009 to June 2010. The 'Your account' section displays the current balance of €226,94 and upcoming allocations of €206,35. The left sidebar contains links for 'Your features', 'Your Members', and 'News'.

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

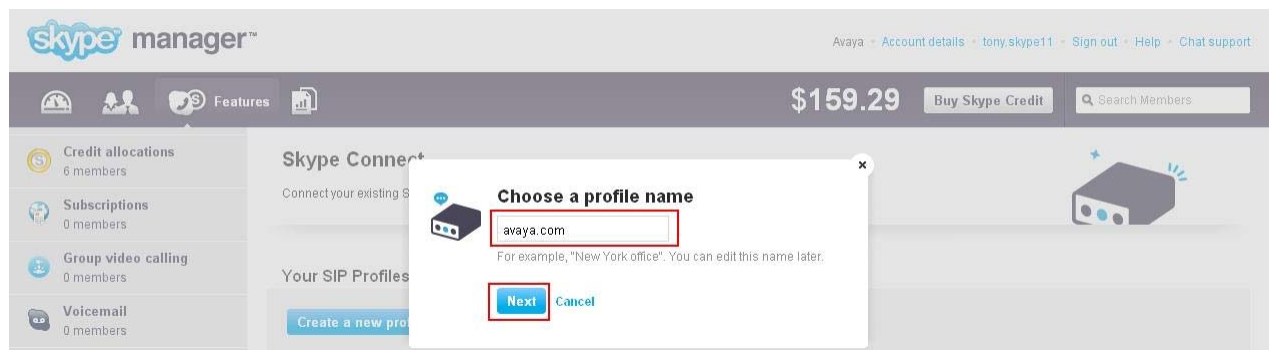
The screenshot shows the Skype Manager Features page. The top navigation bar includes 'Features' (highlighted with a red box) and a toolbar with the current balance of \$159.29, a 'Buy Skype Credit' button, and a search bar. The left sidebar contains a list of features: 'Credit allocations', 'Subscriptions', 'Group video calling', 'Voicemail', 'Online Numbers', 'Call forwarding', and 'Skype Connect' (highlighted with a red box). The main content area displays a table of credit allocations for various members.

☐	Name	Credit	Auto-recharge
☐	Avaya.SILWestminster2 Avaya.SILWestmins	\$0.00	Auto-recharge disabled
☐	Avaya.SILWestminster Avaya.SILWestminst	\$0.00	Auto-recharge disabled
☐	BR Test	\$0.00	Auto-recharge disabled
☐	BR Test2	\$5.84	Auto-recharge disabled
☐	cc.avaya.com	\$0.00	Auto-recharge disabled
☐	dublinsil1	\$0.00	Auto-recharge disabled
☐	dublinsil2	€6,22	Auto-recharge disabled

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

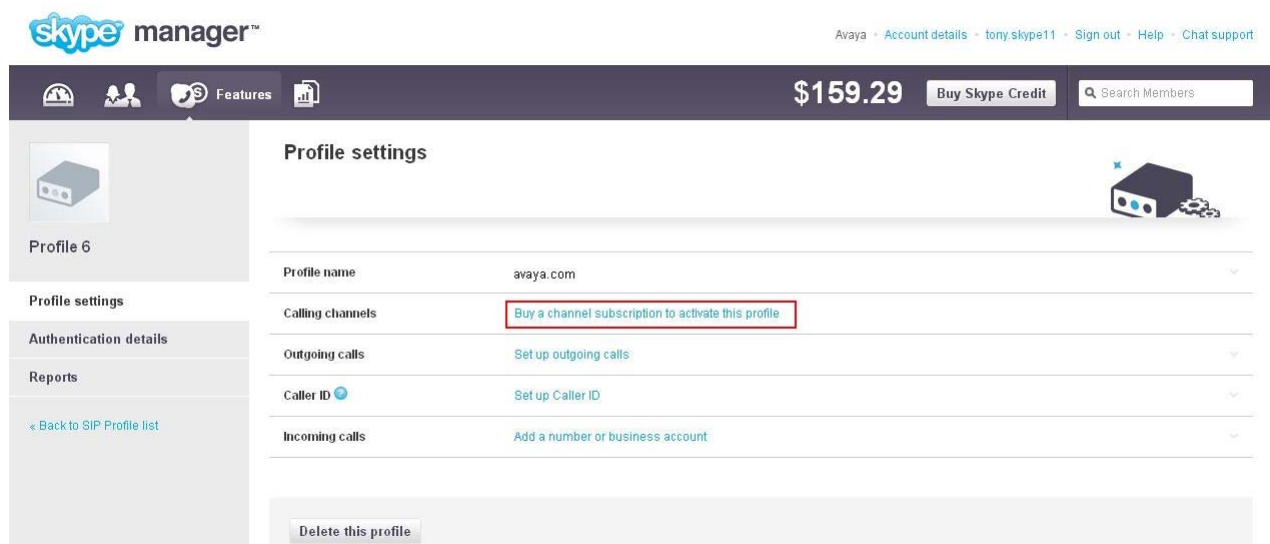


The **Choose a profile name** pop-up window is displayed. Type the name of your SIP Profile and click **Next**.



6.3. Administer Maximum Simultaneous Calls

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



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

 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** (not shown).

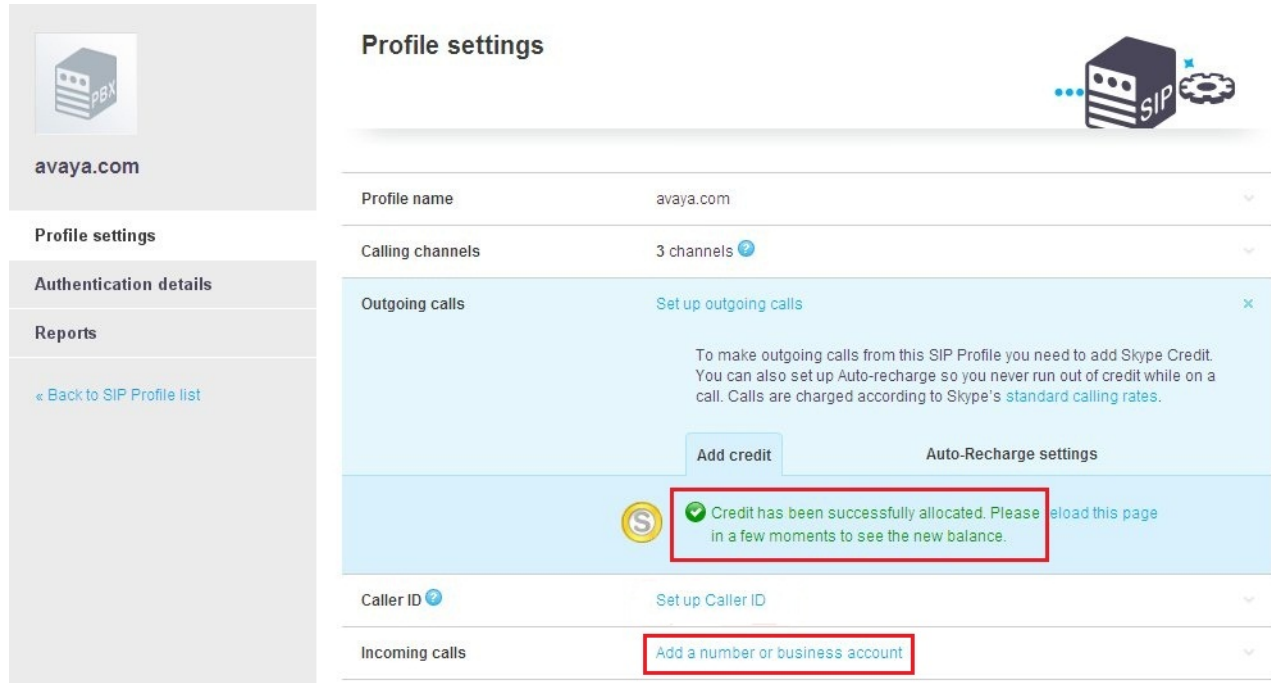
The screenshot shows the Skype Manager interface. The top navigation bar includes the Skype Manager logo, user information (Avaya, Account details, tony.skype11, Sign out, Help, Chat support), and account balance (\$159.29). The left sidebar lists various features: Credit allocations (6 members), Subscriptions (0 members), Group video calling (0 members), Voicemail (0 members), Online Numbers (3 members), Call forwarding (0 members), and Skype Connect (6 profiles). The main content area is titled "Skype Connect" and contains a green success message: "Subscription changed successfully for Profile avaya.com. The profiles updated subscription setting will be available in a few minutes." Below this message is a section for "Your SIP Profiles" with a "Create a new profile" button and a link to "avaya.com View profile".

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 "Profile settings" page for the profile "avaya.com". The left sidebar lists the settings: Profile settings, Authentication details, and Reports, with a link to "Back to SIP Profile list". The main content area shows the "Profile name" as "avaya.com", "Calling channels" as "2 channels", and "Outgoing calls" as "€3,72 Auto-recharge disabled". Below this is the "Auto-recharge settings" section, which includes an "Add credit" button, a text input field containing "\$ 14.17", and another "Add credit" button. A "Take back credit" link is also visible.

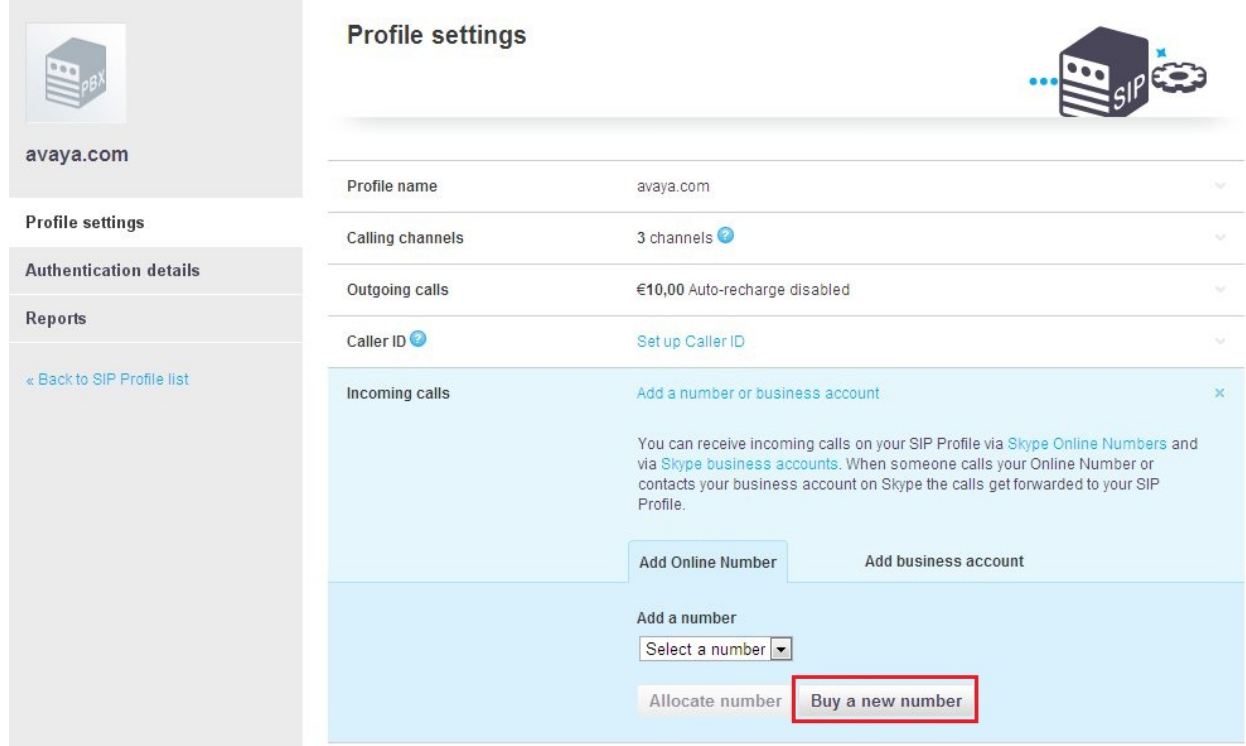
6.4. Administer Online Numbers

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 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 has a header with a SIP icon and a 'Profile settings' title. Below this, there are several sections: 'Profile name' (avaya.com), 'Calling channels' (3 channels), 'Outgoing calls' (Set up outgoing calls), 'Caller ID' (Set up Caller ID), and 'Incoming calls' (Add a number or business account). A confirmation message is displayed in a light blue box: 'Credit has been successfully allocated. Please reload this page in a few moments to see the new balance.' The message is accompanied by a green checkmark icon and a 'Credit' label. The 'Add a number or business account' link in the 'Incoming calls' section is highlighted with a red box.

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



The screenshot shows the 'Profile settings' page for 'avaya.com' with the 'Incoming calls' tab expanded. The left sidebar is the same as in the previous screenshot. The main content area shows the 'Incoming calls' section with the title 'Add a number or business account'. Below this, there is a description: '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'. Below these buttons, there is a section for 'Add a number' with a dropdown menu labeled 'Select a number' and a button labeled 'Allocate number'. The 'Buy a new number' button is 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**.

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 Add a number or business account

Delete this profile

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

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile settings

Profile name avaya.com

Calling channels 3 channels

Outgoing calls €10,00 Auto-recharge disabled

Caller ID Set up Caller ID

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

6.5. 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 an Online Number**.



avaya.com

Profile settings

Authentication details

Reports

[« Back to SIP Profile list](#)

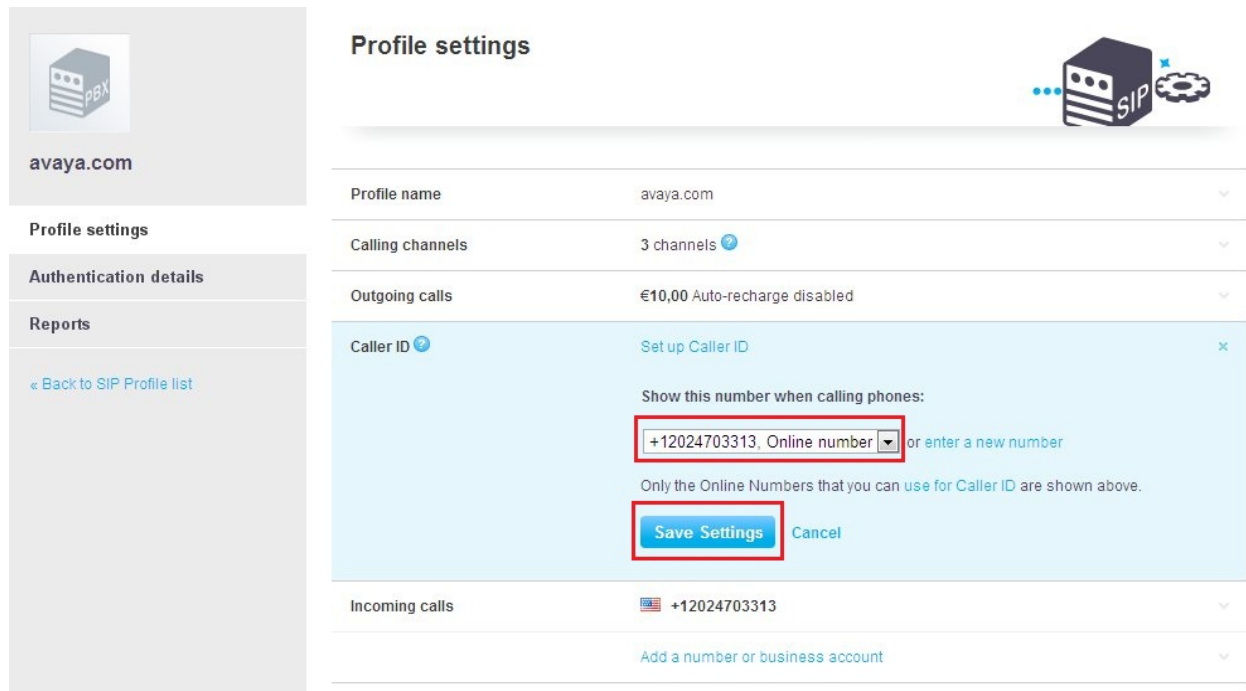
Profile settings



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>Use an Online Number Use a landline number</div>	
Incoming calls	 +12024703313	▼
	Add a number or business account	▼

[Delete this profile](#)

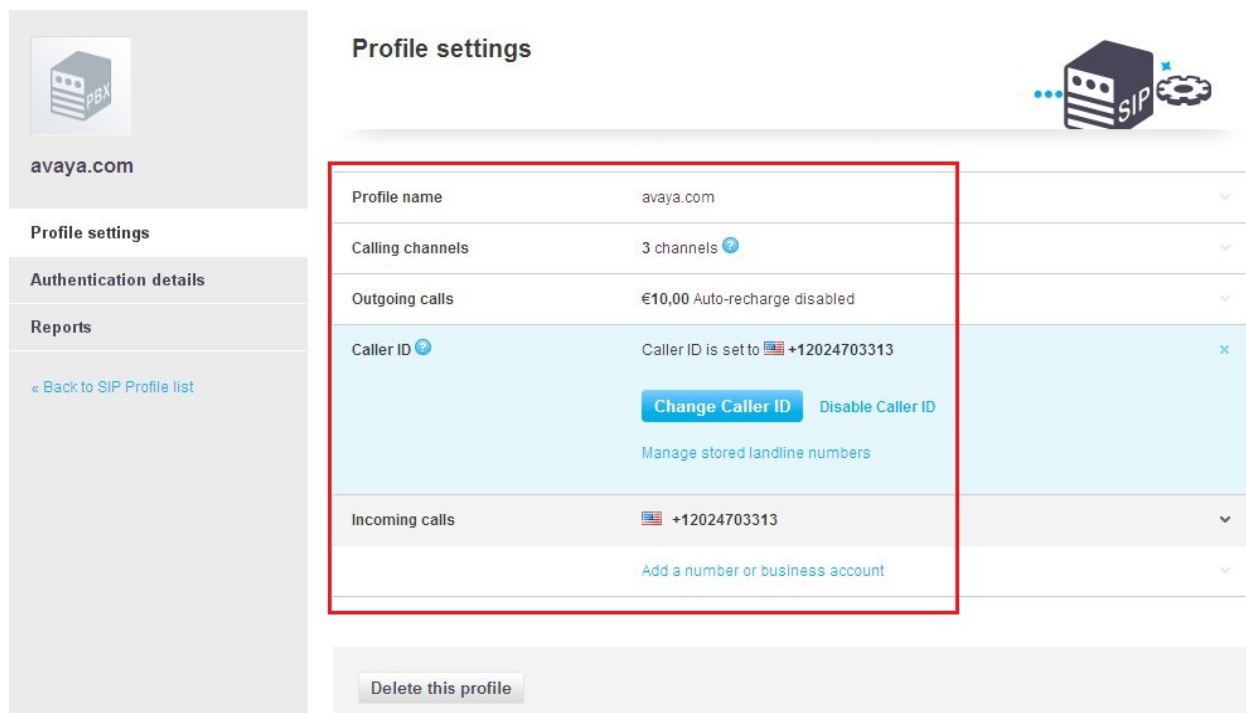
Select an Online Number from the **Show this number when calling phones:** drop-down list box than click **Save Settings**.



The screenshot shows the 'Profile settings' page for a profile named 'avaya.com'. The left sidebar contains navigation links: 'Profile settings' (selected), 'Authentication details', and 'Reports'. A link '« Back to SIP Profile list' is also present. The main content area is titled 'Profile settings' and features a SIP icon. The settings are as follows:

- Profile name: avaya.com
- Calling channels: 3 channels
- Outgoing calls: €10,00 Auto-recharge disabled
- Caller ID: A light blue box contains the text 'Set up Caller ID' and 'Show this number when calling phones:'. Below this is a dropdown menu showing '+12024703313, Online number' and a link 'or enter a new number'. A note states: 'Only the Online Numbers that you can use for Caller ID are shown above.' At the bottom of this box are 'Save Settings' and 'Cancel' buttons.
- Incoming calls: A dropdown menu showing a US flag and '+12024703313', with a link 'Add a number or business account' below it.

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



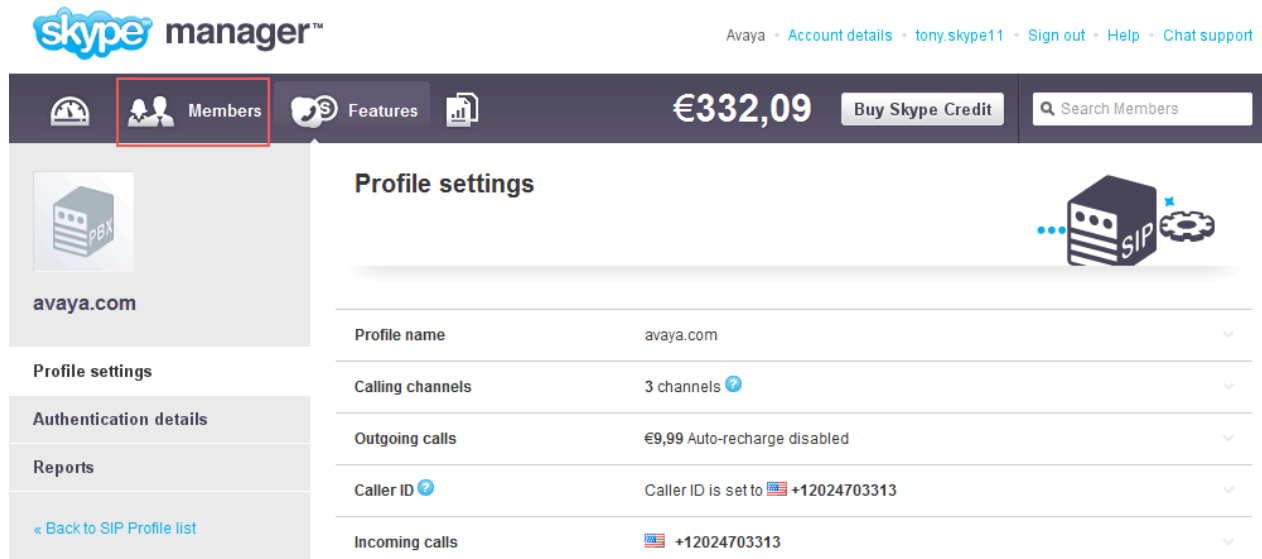
This screenshot shows the 'Profile settings' page with the configuration completed. A red box highlights the 'Caller ID' and 'Incoming calls' sections. The settings are:

- Profile name: avaya.com
- Calling channels: 3 channels
- Outgoing calls: €10,00 Auto-recharge disabled
- Caller ID: The light blue box now shows 'Caller ID is set to' followed by a US flag and '+12024703313'. Below this are buttons for 'Change Caller ID' and 'Disable Caller ID', and a link 'Manage stored landline numbers'.
- Incoming calls: The dropdown menu shows a US flag and '+12024703313', with a link 'Add a number or business account' below it.

At the bottom of the page, there is a button labeled 'Delete this profile'.

6.6. Create a Business User

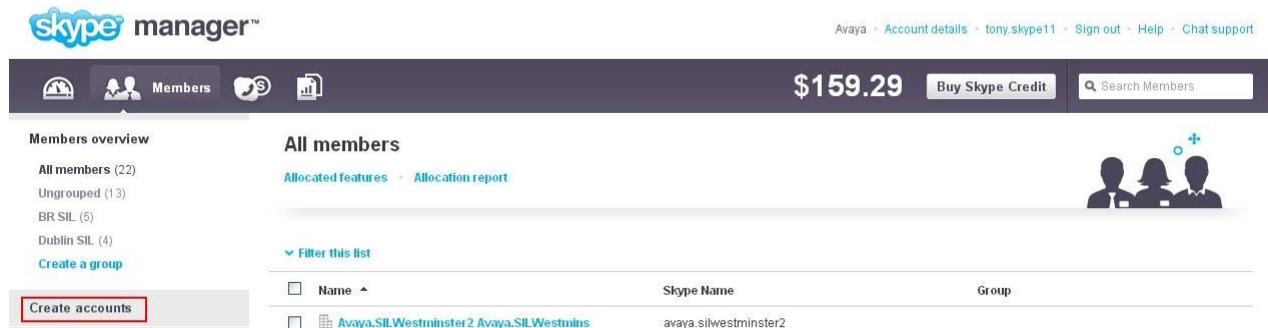
A Business User with a unique Skype Name needs to be created and associated with the extension of the enterprise user (3001) for click-to-call functionality to work from the Skype P2P Network. Click **Members** on the toolbar.



The screenshot shows the Skype Manager interface. The top navigation bar includes the Skype Manager logo, a breadcrumb trail (Avaya > Account details > tony.skype11 > Sign out > Help > Chat support), and a balance of €332,09. The 'Members' tab is highlighted in the toolbar. The left sidebar shows the 'avaya.com' profile and navigation links for Profile settings, Authentication details, Reports, and a link back to the SIP Profile list. The main content area is titled 'Profile settings' and displays a table of account details.

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 **Create accounts**.



The screenshot shows the 'All members' page in Skype Manager. The top navigation bar shows a balance of \$159.29. The 'Members' tab is active. The left sidebar shows a 'Members overview' with links to 'All members (22)', 'Ungrouped (13)', 'BR SIL (5)', 'Dublin SIL (4)', and 'Create a group'. The 'Create accounts' link is highlighted with a red box. The main content area is titled 'All members' and includes links for 'Allocated features' and 'Allocation report'. A table lists the members.

☐	Name	Skype Name	Group
☐	Avaya.SILWestminster2 Avaya.SILWestmins	avaya.silwestminster2	

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

The screenshot shows the 'Create accounts' page in Skype Manager. The top navigation bar includes the Skype Manager logo, user information (Avaya, Account details, tony.skype11), and a balance of \$159.29. The left sidebar shows 'Members overview' and 'Create accounts'. The main content area is titled 'Create accounts' and has three steps: 1. Enter email addresses/Import CSV, 2. Enter details, and 3. Summary. Step 1 is active. It contains a text field for 'Enter email addresses' with 'bob@avaya.com' entered. Below it is a button 'Choose File' and a note: 'The CSV file must contain columns for: First name, Last name and Email address. User name and Password columns can also be added. How to generate a CSV file'. A 'Next' button is at the bottom.

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.

Click **Create accounts**.

The screenshot shows the 'Create accounts' page in Skype Manager, Step 2: Enter details. The top navigation bar is the same as the previous screenshot. The left sidebar shows 'Members overview' and 'Create accounts'. The main content area is titled 'Create accounts' and has three steps: 1. Enter email addresses, 2. Enter details, and 3. Summary. Step 2 is active. It shows a message: 'We found 1 valid email address in your input'. Below this is a section for 'Email address*' and 'Skype Name*'. The email address is 'bob@avaya.com' and the Skype Name is 'Bob.avaya.com'. Below these are fields for 'First name' (Bob), 'Last name' (smith), 'Password' (masked with asterisks), and 'Repeat password' (masked with asterisks). A note says: 'The password needs to be at least 6 characters long and contain at least 1 number.' There is a 'Remove this account' link. At the bottom, there is a section 'Add another account' and a dropdown menu for 'Add members to a group after their account is created' with 'Ungrouped' selected. A 'Create accounts' button is at the bottom.

6.7. Add Business User to SIP Profile

Click **Dashboard** and scroll down and click **Skype Connect**.

Dashboard **\$159.29** [Buy Skype Credit](#)

Reports **Your account** **Account status: Attention required**

Allocations

500
250
0

May 2010 Jun Jul Aug Sep Oct Nov Dec Jan Feb Mar Apr May 2011

Current balance **\$159.29** Auto-recharge is disabled

Upcoming allocations (next 30 days) **\$198.10** [Review allocations](#)

Credit missing \$38.80
[Buy more Skype Credit](#)

Your features **Your members** **News**

6 members have **Skype Credit**

Set up **Subscriptions** for your members

Set up **Group video calling** for your members

3 members have **Online Numbers**

Set up **Call forwarding** for your members

Set up **Voicemail** for your members

5 profiles set up for **Skype Connect**

Your Skype Manager has 22 members
[Create accounts](#)

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

Welcome! Skype Manager is a brand new product that replaces the Business Control Panel. From setting up employee accounts, to allocating Skype Credit, to activating Online Numbers, Skype Manager's completely redesigned interface is much easier to use. Don't forget to check the [Skype Support site](#) for help if you get stuck, and the [Skype Manager forum](#) to share useful information, feedback...

[More news](#)

The **Skype Connect** page is displayed. Click **View profile**.

skype manager™ Avaya [Account details](#) [tony.skype11](#) [Sign out](#) [Help](#) [Chat support](#)

Features

Credit allocations 6 members

Subscriptions 0 members

Group video calling 0 members

Voicemail 0 members

Online Numbers 3 members

Call forwarding 0 members

Skype Connect 5 profiles

Skype Connect

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

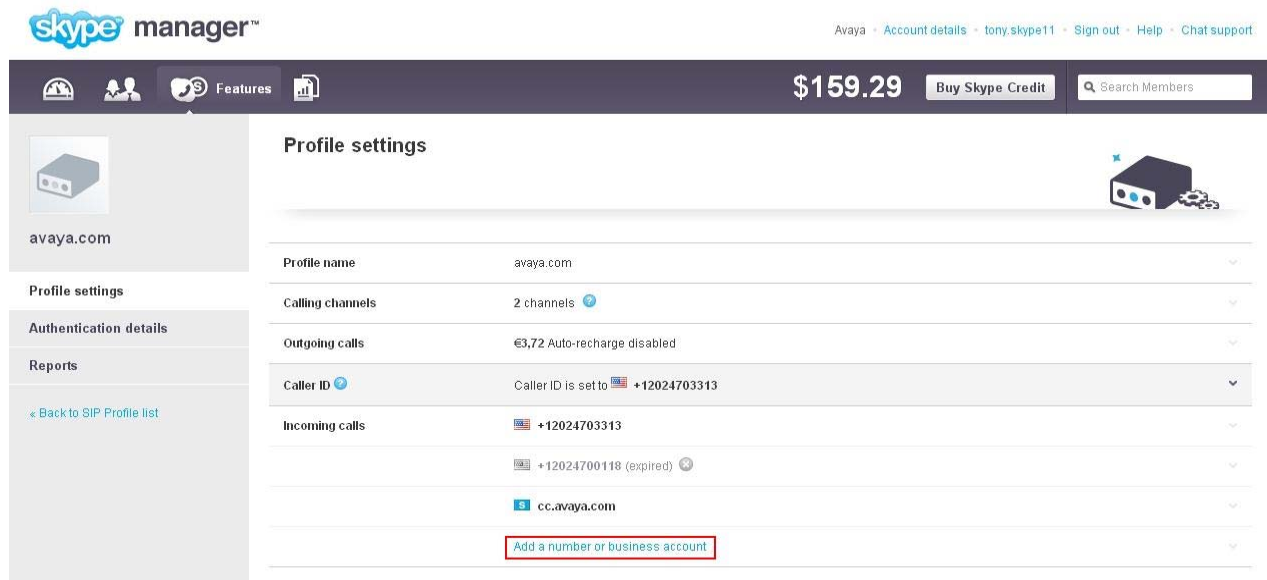
Your SIP Profiles

[Create a new profile](#)

avaya.com [View profile](#)

Channels	2 channels
Outgoing calls	€3,72 available Auto-recharge disabled
Incoming calls	2 Online Numbers 1 Skype account

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



skype manager™

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

\$159.29 Buy Skype Credit Search Members

avaya.com

Profile settings

Authentication details

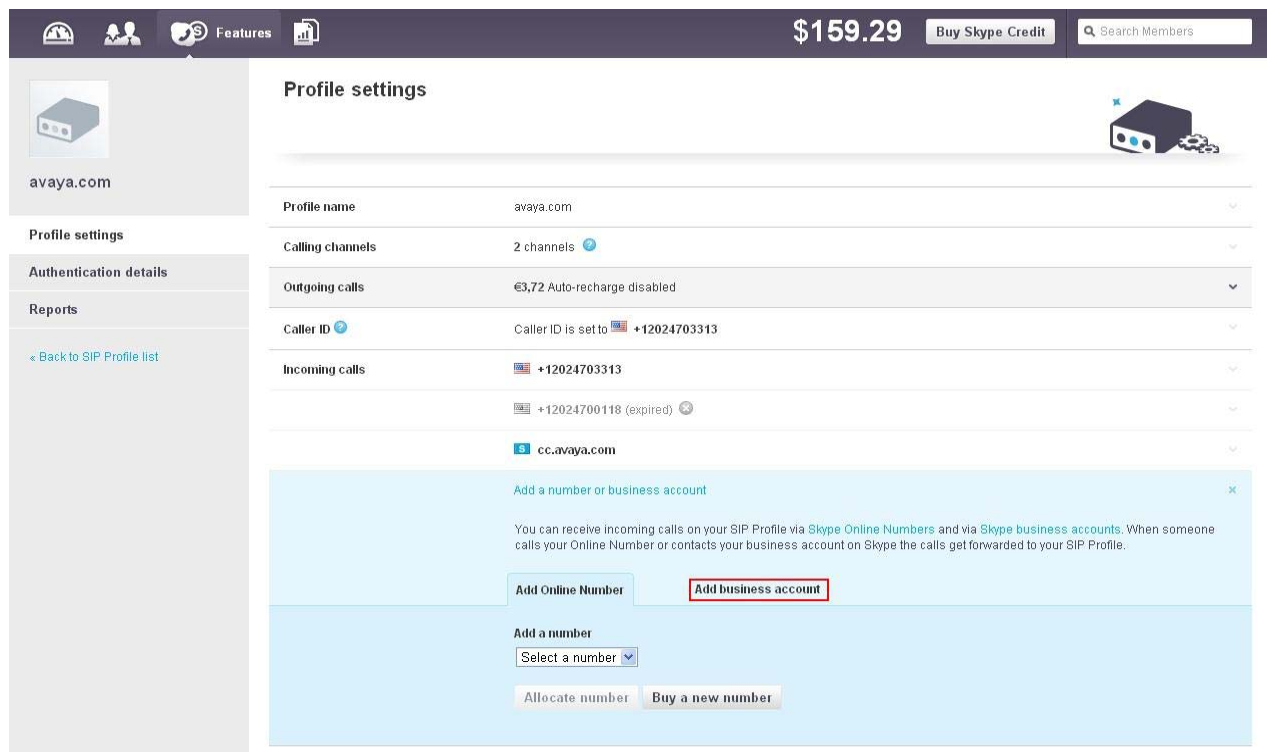
Reports

« Back to SIP Profile list

Profile settings

Profile name	avaya.com
Calling channels	2 channels
Outgoing calls	€3,72 Auto-recharge disabled
Caller ID	Caller ID is set to +12024703313
Incoming calls	+12024703313
	+12024700118 (expired)
	cc.avaya.com
	Add a number or business account

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



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\$159.29 Buy Skype Credit Search Members

avaya.com

Profile settings

Authentication details

Reports

« Back to SIP Profile list

Profile settings

Profile name	avaya.com
Calling channels	2 channels
Outgoing calls	€3,72 Auto-recharge disabled
Caller ID	Caller ID is set to +12024703313
Incoming calls	+12024703313
	+12024700118 (expired)
	cc.avaya.com

[Add a number or business account](#)

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

[Add Online Number](#) [Add business account](#)

Add a number

Select a number

[Allocate number](#) [Buy a new number](#)

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

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

Click **Confirm**.

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

Profile name	avaya.com
Calling channels	2 channels
Outgoing calls	€3,72 Auto-recharge disabled
Caller ID	Caller ID is set to +12024703313
Incoming calls	<div>+12024703313</div> <div>+12024700118 (expired)</div> <div>cc.avaya.com</div> <div>bob.avaya.com</div> <div>Add a number or business account</div>

7. Configure Avaya Aura® Session Border Controller

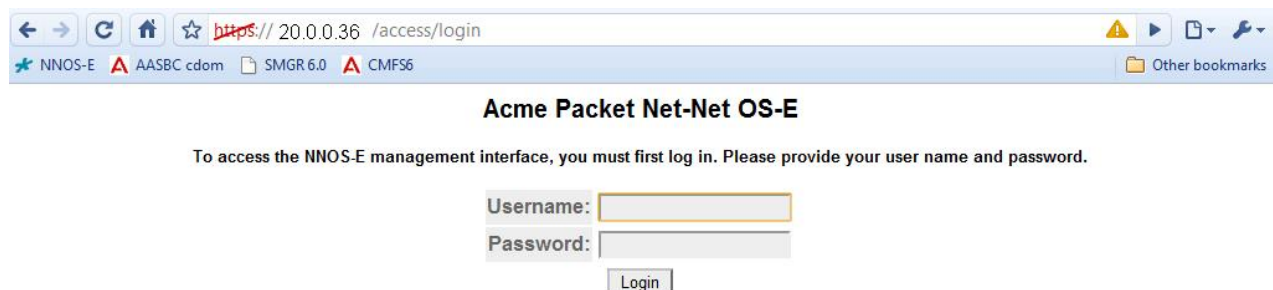
This section provides the procedures for configuring Avaya Aura® Session Border Controller and includes the following items:

- Log in to Avaya Aura® Session Border Controller using the GUI
- Administer SIP Domains
- Administer TLS Certificate
- Administer TLS on the Outside Interface
- Administer Authentication
- Administer Outbound SIP Header Manipulation Rules
- Administer Inbound SIP Header Manipulation Rules
- Administer SIP Header Rules for Topology Hiding
- Save the Configuration

Appendix B contains a sample configuration file in text form.

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

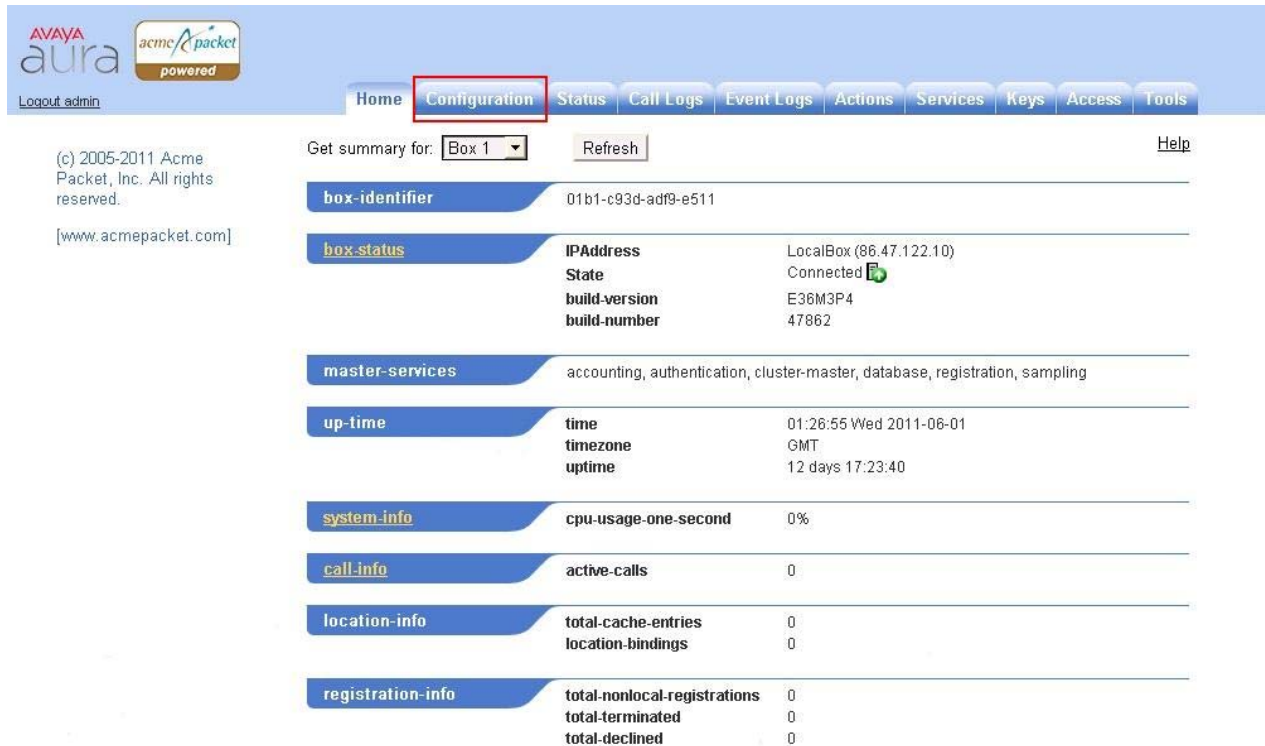
Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® Session Border Controller, using the URL “**https://<ip-address>**”, where “<ip-address>” is the IP address of the inside interface of the Avaya Aura® Session Border Controller. Log in with the appropriate credentials.



The screenshot shows a web browser window with the address bar displaying `https://20.0.0.36/access/login`. The browser's bookmark bar includes entries for NNOS-E, AASBC cdom, SMGR 6.0, and CMFS6. The main content area of the browser shows the title "Acme Packet Net-Net OS-E" and a message: "To access the NNOS-E management interface, you must first log in. Please provide your user name and password." Below this message 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.



AVAYA aura acmepacket powered

Logout admin

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

Get summary for: **Box 1** Refresh Help

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[www.acmepacket.com]

box-identifier	01b1-c93d-adf9-e511	
box-status	IPAddress	LocalBox (86.47.122.10)
	State	Connected 
	build-version	E36M3P4
	build-number	47862
master-services	accounting, authentication, cluster-master, database, registration, sampling	
up-time	time	01:26:55 Wed 2011-06-01
	timezone	GMT
	uptime	12 days 17:23:40
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 PBX [Show advanced](#)

SetResetBackCopyDelete

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

general:

* name

PBX

admin

enabled

(Resource is active)

domain

avaya.com

failover-detection

ping

(Use OPTIONS to detect failures)

servers:

server-pool

[Delete](#)

TPN:Reviewed
SPOC 08/01/2011

Solution & Interoperability Test Lab Application Notes
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54 of 88
Skype-Avaya-TLS

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 TLS Certificate

Select **tls** on the left pane. The **Configure vsp\tls** page is displayed. Select **Add certificate**.

Configuration

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

Configure vsp\tls Show advanced Help Index

Set Reset Back Delete

default-outgoing-settings Configure

default-ca (Delete) ca-file

	FileName	PassphraseTag
Edit Delete	/cxc/certs/sipca.pem	

Add ca-file

default-cert Configure

certificate

	certificate	certificate-file	passphrase-tag	allow-ssl2	allow-ssl3	allow-tls1	allow-null-cipher	dynamic-buffers	enable-cbc-counter-measure	tx-record-length	peer-certificate
Edit Delete	certificate ws-cert	/cxc/certs/ws.cert		false	true	true	disabled	enabled	true	2048	none
Edit Delete	certificate Sk	/cxc/certs/Sk		false	true	true	disabled	enabled	true	2048	none
Edit Delete	certificate aasbc.p12	/cxc/certs/aasbc.p12	aasbc-cert-tag	false	true	true	disabled	enabled	true	2048	none

Add certificate

Set Reset Back

In the next screen, enter a name and select **Create**.

Configuration

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

Create vsp\tls\certificate - Step 1 of 1: Edit certificate Help Index

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

general:

* name Skype

Create Reset Cancel

In the next screen select **Browse System Files** and select the Skype TLS certificate (not shown). Leave all other properties as default. Click **Set** to save.

The screenshot shows the Avaya Aura Configuration interface. The top navigation bar includes 'Home', 'Configuration', 'Status', 'Call Logs', 'Event Logs', 'Actions', 'Services', 'Keys', 'Access', and 'Tools'. The left sidebar shows a tree view of configuration objects: 'cluster' (with sub-item 'box:aaasbc.avaya.com'), 'vsp' (with sub-items 'default-session-config', 'tls', 'session-config-pool', 'dial-plan', 'enterprise', 'accounting', 'dns', and 'settings'), and 'settings'. The main content area is titled 'Configure vsp/tls/certificate test' and includes a 'Show advanced' link and a 'Help Index' link. Below the title are buttons for 'Set', 'Reset', 'Back', 'Copy', and 'Delete'. The 'Set' button is highlighted with a red box. The configuration is divided into two sections: 'general' and 'other properties'. The 'general' section contains fields for 'name' (Skype), 'certificate-file' (/cxc/certs/Skype), and 'passphrase-tag'. The 'certificate-file' field has a 'Browse System Files' button highlighted with a red box. The 'other properties' section contains several dropdown menus: 'allow-ssl2' (false), 'allow-ssl3' (true), 'allow-tls1' (true), 'allow-null-cipher' (disabled), 'dynamic-buffers' (enabled), and 'enable-chc-counter-measure' (true). The 'allow-null-cipher' and 'dynamic-buffers' fields have additional text indicating their status: '(Resource is inactive)' and '(Resource is active)' respectively.

Configuration: all

Configuration Setup View

cluster

- box:aaasbc.avaya.com

vsp

- default-session-config
- tls
- session-config-pool
- dial-plan
- enterprise
- accounting
- dns
- settings

Configure vsp/tls/certificate test Show advanced Help Index

Set Reset Back Copy Delete

general:

* name Skype

certificate-file /cxc/certs/Skype Browse System Files

passphrase-tag Manage Password

other properties:

allow-ssl2 false

allow-ssl3 true

allow-tls1 true

allow-null-cipher disabled (Resource is inactive)

dynamic-buffers enabled (Resource is active)

enable-chc-counter-measure true

7.4. Administer TLS on the outside interface

Select **sip** on the left pane. The **Configure cluster\box:aaasbc.avaya.com\interface eth2\ip outside\sip** page is displayed. Select **Add-tls-port**.

The screenshot shows the Avaya Aura Configuration interface. The left pane displays a tree view with 'sip' selected under 'ip outside'. The main pane shows the configuration for 'Configure cluster\box:aaasbc.avaya.com\interface eth2\ip outside\sip'. The 'admin' dropdown is set to 'enabled'. The 'nat-translation' dropdown is set to 'disabled'. The 'nat-add-received-from' dropdown is set to 'disabled'. The 'nat-add-X-Remote-Info' dropdown is set to 'enabled'. The 'load-balance-head-end' dropdown is set to 'false'. The 'udp-port' table has one row with 'udp-port' 5060. The 'tcp-port' and 'tls-port' sections have 'Add tcp-port' and 'Add tls-port' links respectively. The 'certificate' dropdown is set to 'vsp/tls/certificate ws-cert'.

udp-port	from-server	to-server	transport	remote-port	certificate
udp-port 5060	Edit	Edit	any	0	Edit

In the next screen, enter **5061** for the port number and select **Create**.

The screenshot shows the Avaya Aura Configuration interface. The left pane displays a tree view with 'sip' selected under 'ip outside'. The main pane shows the configuration for 'Create cluster\box 1\interface eth2\ip outside\sip\tls-port 5061 - Step 1 of 1: Edit tls-port 5061'. The 'port' field is set to '5061'. The 'Create' button is highlighted.

* port
5061

A TLS certificate now needs to be associated with the newly created TLS port. In the right-hand pane, select **Create** to create an association to the TLS certificate administered in **Section 7.3**.

Configuration

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

Status Summary Logout admin

Configuration: all

Configuration Setup View

cluster

- box:aaasbc.avaya.com
 - interface eth0
 - interface eth2
 - ip outside
 - ssh
 - web
 - sip**
 - icmp
 - media-ports
 - routing
 - kernel-filter
 - cli
- vsp
 - default-session-config
 - tls
 - session-config-pool
 - dial-plan
 - enterprise
 - accounting
 - dns
 - settings

Configure cluster:aaasbc.avaya.com/interface eth2/ip outside/sip Help Index

Set Reset Back Delete

Press "Set" to keep these values.

admin	enabled	(Resource is active)													
nat-translation	disabled	(Resource is inactive)													
nat-add-received-from	disabled	(Resource is inactive)													
nat-add-X-Remote-Info	enabled	(Resource is active)													
load-balance-head-end	false														
udp-port	<table border="1"> <thead> <tr> <th>udp-port</th> <th>from-server</th> <th>to-server</th> <th>transport</th> <th>remote-port</th> <th>certificate</th> </tr> </thead> <tbody> <tr> <td>Edit Delete</td> <td>udp-port 5060</td> <td>Edit</td> <td>Edit</td> <td>any</td> <td>0</td> <td>Edit</td> </tr> </tbody> </table>	udp-port	from-server	to-server	transport	remote-port	certificate	Edit Delete	udp-port 5060	Edit	Edit	any	0	Edit	Add udp-port
udp-port	from-server	to-server	transport	remote-port	certificate										
Edit Delete	udp-port 5060	Edit	Edit	any	0	Edit									
tcp-port	Add tcp-port														
tls-port	<table border="1"> <thead> <tr> <th>tls-port</th> <th>from-server</th> <th>to-server</th> <th>transport</th> <th>remote-port</th> <th>certificate</th> </tr> </thead> <tbody> <tr> <td>Edit Delete</td> <td>tls-port 5061</td> <td>Edit</td> <td>Edit</td> <td>any</td> <td>0</td> <td>Edit</td> </tr> </tbody> </table>	tls-port	from-server	to-server	transport	remote-port	certificate	Edit Delete	tls-port 5061	Edit	Edit	any	0	Edit	Add tls-port
tls-port	from-server	to-server	transport	remote-port	certificate										
Edit Delete	tls-port 5061	Edit	Edit	any	0	Edit									
certificate	vsp/tls/certificate ws-cert Edit Create														

In the next screen, enter the path to the TLS certificate administered in **Section 7.3** and select **Create**.

Configuration

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

Status Summary Logout admin

Configuration: all

Configuration Setup View

cluster

- box:aaasbc.avaya.com
 - interface eth0
 - interface eth2
 - ip outside
 - ssh
 - web
 - sip**
 - icmp
 - media-ports
 - routing
 - kernel-filter
 - cli
 - vsp
 - default-session-config
 - tls
 - session-config-pool
 - dial-plan
 - enterprise
 - accounting
 - dns
 - settings

Enter a path for: certificate. Example is vsp/tls/certificate test

Path: vsp/tls/certificate Skype

Create Back

One will then be returned to the previous screen, select **Set** to save.

7.5. Administer Authentication

Select **sip-gateway Telco** on the left pane. The **Configure vsp\enterprise\servers\sip-gateway Telco** page is displayed. Click on the **Show advanced** button.

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](#)

Enter the following details under **registration-failover**:

- **User** The SIP User number assigned to the avaya.com SIP profile upon creation in Skype Manager. The number will be seen in the Authentication Details section of the SIP user profile.
- **password-tag** i.e. **avayapass**
- **add-user-to-contact** **enabled**

Configure vsp\enterprise\servers\sip-gateway SkypeAMS [Show basic](#) [Help](#) [Index](#)

[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
peer-identity	sip.skype.com
admin	enabled (Resource is active)
domain	sip.skype.com
directory	Create
failover-detection	register (Use REGISTER to detect failures)

registration-failover:

user	99051000109702
password-tag	avayapass Manage Password
add-user-to-contact	enabled (Resource is active)

7.6. 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 acmePacket Configuration page. The left navigation pane is expanded to show the path: **vsp** → **session-config-pool** → **entry ToTelco** → **to-uri-specification**. The right pane displays the 'Configuration Loaded' message: 'The configuration has been successfully loaded.'

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

The screenshot shows the acmePacket Configuration page with the configuration form for 'to-uri-specification' under 'entry ToTelco'. The form has the following fields:

Field	Value	Notes
user	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
host	next-hop-domain	(Net-Net OS-E uses the domain of the next-hop server.)
port	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
display	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
transport	to-uri	(Net-Net OS-E uses the value from the incoming TO URI.)
user-param	omit	
user-truncate-non-digits	disabled	(Resource is inactive)
uri-parameter	Add uri-parameter	

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 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 <input type="text" value="99051000109702"/> or select from <input type="text" value="99051000109702"/>
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.)
display	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.)
user-agent-aware-display-translation	<input type="text" value="disabled"/> (Resource is inactive)
transport	<input type="text" value="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 <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="request-uri"/> or select from <input type="text" value="request-uri"/> (Net-Net OS-E uses the value from the incoming REQUEST URI.)
transport	<input type="text" value="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

7.7. 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 'Configuration: all' with 'request-uri-specification' selected. The main pane is titled 'Configure vsp\session-config-pool\entry ToPBX\request-uri-specification'. It has buttons for 'Set', 'Reset', 'Back', and 'Delete'. The configuration table has three rows: 'user' with a text input 'request-uri' and a dropdown 'request-uri'; 'host' with a text input 'next-hop-domain' and a dropdown 'next-hop-domain'; and 'port' with a text input 'request-uri' and a dropdown 'request-uri'. The 'host' dropdown is highlighted with a red box.

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 'Configuration: all' with 'p-asserted-identity-uri-specification' selected. The main pane is titled 'Configure vsp\session-config-pool\entry ToPBX\p-asserted-identity-uri-specification'. It has buttons for 'Set', 'Reset', 'Back', and 'Delete'. The configuration table has three rows: 'user' with a text input 'request-uri' and a dropdown 'request-uri'; 'host' with a text input 'next-hop-domain' and a dropdown 'next-hop-domain'; and 'port' with a text input 'same-uri' and a dropdown 'same-uri'. The 'host' dropdown is highlighted with a red box.

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.)

7.8. 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 Session Manager configuration interface. On the left, the 'Configuration: all' tree view is expanded to 'vsp' > 'pre-session-config' > 'sip-header-settings'. The 'sip-header-settings' node is highlighted with a red box. On the right, the 'Configure vsp\pre-session-config\sip-header-settings' page is displayed. It features a table with two rows: 'admin' with a value of 'enabled' and '(Resource is active)', and 'rule' with a value of 'Add rule'. The 'Add rule' link is highlighted with a red box. Below the table are buttons for 'Set', 'Reset', and 'Back', and links for 'Help' and 'Index'.

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

The screenshot shows the Avaya Aura Session Manager configuration interface. On the left, the 'Configuration: all' tree view is expanded to 'vsp' > 'pre-session-config' > 'sip-header-settings'. The 'sip-header-settings' node is highlighted with a red box. On the right, the 'Create vsp\pre-session-config\sip-header-settings\rule - Step 1 of 1: Edit rule' page is displayed. It features a text input field for '* name' with the value 'P-Site'. Below the input field are buttons for 'Create', 'Reset', and 'Cancel'. The 'Create' button is highlighted with a red box.

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

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](#)

7.9. Save the Configuration

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

The screenshot shows a web-based configuration interface. On the left, under 'Configuration: all', there is a menu with 'Configuration', 'Setup', and 'View' tabs. The 'Configuration' tab is active, and a sub-menu is open showing options: 'Update and save configuration' (highlighted with a red box), 'Reload configuration', 'Validate configuration', 'Analyze configuration', 'Search configuration', 'Save as XML', and 'Load from XML'. Below this are expandable sections for 'session-config-pool', 'dial-plan', 'registration-plan', 'enterprise', 'accounting', 'dns', 'settings', and 'services-routing'. The main panel is titled 'Configure vsp\pre-session-config\sip-header-settings\rule P-Site' with links for 'Help' and 'Index'. It contains buttons for 'Set', 'Reset', 'Back', 'Copy', and 'Delete'. The configuration fields are: '* name' (P-Site), 'description' (empty), 'condition' (with '* condition-type' set to 'match-header' and '* name' set to 'P-Site'), and 'action' (with '* action-type' set to 'strip-header'). At the bottom, there are 'Set', 'Reset', 'Back', and 'Copy' buttons, along with 'Help' and 'Index' links.

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

This screenshot shows the same configuration interface as the previous one, but with a red box highlighting a message at the top: 'Configuration Updated and Saved'. Below the message, it says 'The running configuration has been updated and saved.' The rest of the interface, including the left-hand menu and the main configuration fields for the 'P-Site' rule, remains the same as in the previous screenshot.

8. General Test Approach and Test Results

The test environment consisted of:

- A simulated enterprise with Avaya Aura[®] System Manager, Avaya Aura[®] Session Manager, Avaya Aura[®] Communication Manager, Avaya phones and Avaya Aura[®] Session Border Controller.
- A Beta 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.
- Long duration calls.

9. Verification Steps

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

The screenshot shows the Avaya Aura Session Border Controller GUI. The top navigation bar includes links for Home, Configuration, Status, Call Logs (highlighted), Event Logs, Actions, Services, Keys, Access, and Tools. The main content area displays a summary for a selected box (Box 1) with a Refresh button and a Help link. The summary is organized into several sections:

- box-identifier**: 01b1-c93d-adf9-e511
- box-status**:
 - IPAddress: LocalBox (86.47.122.10)
 - State: Connected (green icon)
 - build-version: E36M3P4
 - build-number: 47862
- master-services**: accounting, authentication, cluster-master, database, registration, sampling
- up-time**:
 - time: 01:26:55 Wed 2011-06-01
 - timezone: GMT
 - uptime: 12 days 17:23:40
- 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 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).

The screenshot shows the 'Sessions' page in the Avaya Aura Session Manager interface. The page has a blue header with the 'acme packet' logo and 'Call Logs' title. Below the header is a navigation bar with links: Home, Configuration, Status, Call Logs, Event Logs, Actions, Services, Keys, Access, Tools. The 'Sessions' page is selected. On the left, there is a 'Select:' menu with options: Sessions, User Sessions, Devices, SIP Messages, H323 Messages, Accounting Calls, Monitored URIs, Monitored Calls, Files, and Database Archives. The main content area shows a search bar with 'Search Type: All Sessions' and a 'Search' button. Below the search bar, it says 'View All Sessions'. The session list is displayed with columns: Created, Method, Result, From, To, Call ID, and Events. The first session is highlighted with a 'Detail' link. The session details are: Created: 12:39:58.780 Mon 2010-06-14, Method: INVITE, Result: sip:+12024702967@avaya.com, To: sip:0035312075630@avaya.com, Call ID: CXC-208-5c4efc30-ecdd78c1-13c4-4c16231e-66b3c6c0-4ebec4cc. The page footer shows '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 Session Border Controller and the 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 R6.1, Avaya Aura[®] Communication Manager R6.0.1 and Avaya Aura[®] Session Border Controller R6.0 can be configured to interoperate successfully with the Skype Connect service. This solution provides users of Avaya Aura[®] Communication Manager R6.0.1 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. These Application Notes also demonstrated that the Avaya Aura[®] Session Border Controller can interoperate with the Skype Connect service over TLS. 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

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

- [1] *Avaya Aura[®] Session Manager Overview*, Doc # 03603323, Issue 1 Release 6.1
- [2] *Administering Avaya Aura[®] Session Manager*, Doc # 03603324, Issue 1 Release 6.1
- [3] *Maintaining and Troubleshooting Avaya Aura[®] Session Manager*, Doc # 03603325, Issue 1 Release 6.1
- [4] *Administering Avaya Aura[™] Communication Manager*, Doc # 03-300509, Issue 6.0
- [5] *Avaya Aura[™] Session Border Controller System Administration V6.0*,
<https://support.avaya.com/css/P8/documents/100111137>

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

- [6] *Skype Connect product datasheet*, 2011.
<http://www.skype.com/go/skype.connect.product.datasheet>

Appendix A – Skype Button

In **Section 6.6** 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 6.6** and select a Skype Button.

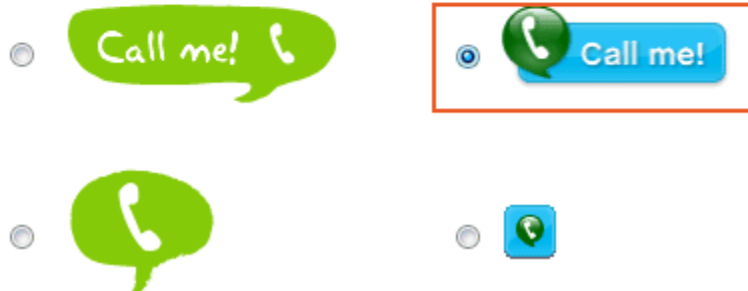
Get a Skype button

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
```

Appendix B – Avaya Aura® SBC Configuration File

```
#
# Copyright (c) 2004-2011 Acme Packet Inc.
# All Rights Reserved.
#
# File: /cxc/cxc.cfg
# Date: 09:58:57 Mon 2011-05-23
#
config cluster
config box 1
set hostname aaasbc.avaya.com
set timezone Etc/GMT
set name aaasbc.avaya.com
set identifier 00:ca:fe:22:12:43
config interface eth0
config ip inside
set ip-address static 20.0.0.36/25
config ssh
return
config snmp
set trap-target 20.0.0.40 162
set trap-filter generic
set trap-filter dos
set trap-filter sip
set trap-filter system
return
config web
return
config web-service
set protocol https 8443
set authentication certificate "vsp\tls\certificate ws-cert"
return
config sip
set tcp-port 5060 "" "" any 0
return
config icmp
return
config media-ports
return
config routing
config route Default
set gateway 20.0.0.1
return
config route Static0
set destination network 192.11.13.4/30
```

```
set gateway 20.0.0.39
return
config route to86
set destination network 86.0.0.0/24
set gateway 20.0.0.1
return
config route Static2
set admin disabled
return
config route Static3
set admin disabled
return
config route Static4
set admin disabled
return
config route Static5
set admin disabled
return
config route Static6
set admin disabled
return
config route Static7
set admin disabled
return
config route Static1
set admin disabled
return
return
return
config ip mgmt
config routing
config route Default
return
config route Static0
return
config route Static1
return
config route Static2
return
config route Static3
return
config route Static4
return
config route Static5
return
config route Static6
```

```

return
config route Static7
return
return
return
return
config interface eth2
config ip outside
set ip-address static 86.47.122.10/25
config ssh
set authentication password
set account root
return
config web
return
config sip
set udp-port 5060 "" "" any 0
set tls-port 5061 "" "" any 0
set certificate vsp\tls\certificate Skype
return
config icmp
return
config media-ports
return
config routing
config route Default
set admin disabled
return
config route 8.8.8.8
set admin disabled
set destination host 8.8.8.8
set gateway 86.47.122.7
return
config route "APKT Mouta"
set admin disabled
set destination host 81.134.16.151
set gateway 86.47.122.7
return
config route SkypeAMS
set destination host 80.252.85.76
set gateway 86.47.122.7
return
return
config kernel-filter
config allow-rule allow-sip-udp-from-peer-1
set destination-port 5060

```

```
set source-address/mask 86.47.122.0/24
set protocol udp
return
config deny-rule deny-all-sip
set destination-port 5060
return
return
return
return
config cli
set prompt aaasbc.avaya.com
return
return
return
```

```
config services
config event-log
config file access
set filter access info
set count 3
return
config file system
set filter system info
set count 3
return
config file errorlog
set filter all error
set count 3
return
config file db
set filter db debug
set filter dosDatabase info
set count 3
return
config file management
set filter management info
set count 3
return
config file peer
set filter sipSvr info
set count 3
return
config file dos
set filter dos alert
set filter dosSip alert
set filter dosTransport alert
```

```
set filter dosUrl alert
set count 3
return
config file krnlsys
set filter krnlsys debug
set count 3
return
return
return
```

```
config master-services
config cluster-master
set group 2
return
config accounting
set group 6
return
config authentication
return
config database
set group 6
set media enabled
return
config registration
return
config sampling
config database
config status
config interface-details
return
config interface-throughput
return
config system-heap
return
config available-memory
return
config trunk-groups
return
config cpu-usage
return
config sip-stack
return
config call-admission-control
return
config dos-transport-counters
return
```

```
config dos-sip-counters
return
config dos-url-counters
return
return
return
return
return

config vsp
set admin enabled
config default-session-config
config media
set anchor enabled
set rtp-stats enabled
return
config sip-directive
set directive allow
return
config log-alert
set apply-to-methods-for-filtered-logs INVITE+REGISTER
return
config third-party-call-control
set admin enabled
set handle-refer-locally disabled
return
return
config tls
config default-ca
set ca-file /cxc/certs/sipca.pem
return
config certificate ws-cert
set certificate-file /cxc/certs/ws.cert
return
config certificate Skype
set certificate-file /cxc/certs/Skype
return
config certificate aasbc.p12
set certificate-file /cxc/certs/aasbc.p12
set passphrase-tag aasbc-cert-tag
return
return
config session-config-pool
config entry toGamma
config to-uri-specification
set host 83.245.6.117
```

```

return
config from-uri-specification
set user 0035391733202
set host local-ip
set display 0035391733202
return
config request-uri-specification
set host 83.245.6.117
return
config p-asserted-identity-uri-specification
set user 0035391733202
set host local-ip
set display 0035391733202
set create true
set use-original-from-header true
return
config header-settings
set pAssert-mode enabled
config reg-ex-header 0
set destination P-Asserted-Identity
set create From .* <sip:0035391733202@83.245.6.117>
return
return
return
config entry SMr61
config to-uri-specification
set host next-hop-domain
set transport TCP
return
config from-uri-specification
set host next-hop-domain
set transport TCP
return
config request-uri-specification
set host next-hop-domain
set port next-hop
set transport TCP
return
config sip-directive
set directive allow
return
config accounting
set target database "vsp\accounting\database\group localdb"
return
config log-alert
return

```

```
return
config entry Discard
config sip-directive
return
return
config entry toSkypeAMS
config to-uri-specification
set host sip.skype.com
set transport omit
return
config from-uri-specification
set user 99051000109702
set host sip.skype.com
set display omit
set transport omit
return
config request-uri-specification
set host sip.skype.com
set transport omit
return
config p-asserted-identity-uri-specification
set host sip.skype.com
set transport omit
return
config contact-uri-settings-out-leg
set add-maddr disabled
return
config media
set anchor enabled
set inactivity-timeout enabled "0 days 01:00:00"
set rtp-stats enabled
return
config in-encryption
set mode offer
set require-tls true
return
config out-encryption
set mode offer
set require-tls true
return
config sip-directive
set directive allow
return
config authentication
set handle-challenge-locally enabled
return
```

```

config log-alert
  set apply-to-methods-for-filtered-logs INVITE+REGISTER
return
return
config entry FromSkypeAMS
  config in-encryption
    set mode offer
    set require-tls true
  return
  config out-encryption
    set require-tls true
  return
  config sip-directive
    set directive allow
  return
  config authentication
    set handle-challenge-locally enabled
  return
  config log-alert
  return
return
return
config dial-plan
config route Default
  set admin disabled
  set priority 500
  set location-match-preferred exclusive
  set session-config vsp\session-config-pool\entry Discard
return
config route toGamma00353
  set admin disabled
  set peer server "vsp\enterprise\servers\sip-gateway Gamma"
  set session-config vsp\session-config-pool\entry toGamma
  set request-uri-match phone-prefix 00353 5
return
config route toSkypeAMS
  set peer server "vsp\enterprise\servers\sip-gateway SkypeAMS"
  set request-uri-match phone-prefix 00 5
return
config route toMX
  set admin disabled
  set peer server "vsp\enterprise\servers\sip-gateway MX"
  set request-uri-match phone-prefix 01306 5
return
config route fromSkypetoSM
  config condition-list

```

```

set operation OR
set request-uri-condition user match 99051000109702
set request-uri-condition user match 12024703313
return
set peer server "vsp\enterprise\servers\sip-gateway SMr61"
set session-config vsp\session-config-pool\entry SMr61
set request-uri-match condition-list
return
return
config enterprise
config servers
config sip-gateway SMr61
set domain avaya.com
set failover-detection ping
set outbound-session-config-pool-entry vsp\session-config-pool\entry SMr61
config server-pool
config server PBX1
set host 20.0.0.20
set transport TCP
return
return
return
config sip-gateway Gamma
set failover-detection ping
set outbound-session-config-pool-entry vsp\session-config-pool\entry toGamma
config server-pool
config server Gamma
set host 83.245.6.117
return
return
return
config sip-gateway SkypeAMS
set peer-identity sip.skype.com
set domain sip.skype.com
set failover-detection register
set user 99051000109702
set password-tag avayapass
set add-user-to-contact enabled
set inbound-session-config-pool-entry vsp\session-config-pool\entry FromSkypeAMS
set outbound-session-config-pool-entry vsp\session-config-pool\entry toSkypeAMS
config server-pool
config server SkypeAMS
set host sip.skype.com
set transport TLS "vsp\tls\certificate Skype"
set port 5061
return

```

```
return
return
config sip-gateway MX
set failover-detection ping
config server-pool
config server MX
set host 20.0.0.10
set transport TCP
return
return
return
return
return
config accounting
config database
config group localdb
config server localdb
set type local
set username postgres
set password-tag dbpassword
return
return
return
return
config dns
config resolver
config server 193.88.6.227
set type dns-only
set name SkypeTestDNS
return
return
config host sip.skype.com
set address 80.252.85.76
return
config service sip.skype.com sip any
set rule 80.252.85.76 5061 0 0
return
return
config settings
set read-header-max 8191
return
return

config external-services
return
```

```
config preferences
config gui-preferences
  set enum-strings RequestURIStringSource 83.245.6.117
  set enum-strings URIStringSource 0035391733202
  set enum-strings GeneralURIStringSource 0035391733202
  set enum-strings DisplayURIStringSource 0035391733202
  set enum-strings SIPSourceHeader P-Asserted-Identity
  set enum-strings RequestURIStringSource sip.skype.com
  set enum-strings URIStringSource 99051000109702
  set enum-strings GeneralURIStringSource sip.skype.com
return
return
config access
config permissions superuser
  set cli advanced
return
config permissions read-only
  set config view
  set actions disabled
return
config users
config user admin
  set password 0x00450df4f8c0700d7cfc8fbc2a173fe7f6b2d630e32e1daed35ba005a2
  set permissions access\permissions superuser
return
config user cust
  set password 0x00eab23aba77bb7c2b3cbcdc78bb39489297c66b9a6e61a262a76625e6
  set permissions access\permissions read-only
return
config user init
  set password 0x00bc74e4e043fb499485f78f14a6a26cc816ad64871b2e5ed8367ec4fa
  set permissions access\permissions superuser
return
config user craft
  set password 0x00ad62ac693367bea4327d1d928775b6a634b28b0f5afce57a4a11be66
  set permissions access\permissions superuser
return
config user dadmin
  set password 0x00be7791ef88476292ea9de8c5d6c6084a5dbaec4eacef5284b04f4c90
  set permissions access\permissions read-only
return
return
return
config features
return
```

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