



Configuring SIP Telephony between Avaya one-X Quick Edition IP Telephones and Avaya Communication Manager with Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe steps to configure SIP connectivity between Avaya one-X Quick Edition IP Telephones and Avaya SIP Enablement Services (SES) without using the Avaya one-X Quick Edition G10 PSTN Gateway. Avaya one-X Quick Edition IP Telephones as SIP identities in the SES Server use a numbering plan for the one-X Quick Edition network to reach endpoints registered to Avaya Communication Manager.

Procedures for the staging of Avaya IP Telephones configured with the Avaya one-X Quick Edition software will be discussed briefly in these Application Notes.

1. Introduction

These Application Notes describe steps to configure SIP connectivity between Avaya one-X Quick Edition IP Telephones and Avaya SIP Enablement Services (SES) without using the Avaya one-X Quick Edition G10 PSTN Gateway. Avaya one-X Quick Edition IP Telephones as SIP identities in the SES Server use a numbering plan for the one-X Quick Edition network to reach endpoints registered to Avaya Communication Manager. The numbering plan configured on Avaya Communication Manager can be used to route calls through the SES Server and forwarded to the one-X Quick Edition network.

The Avaya one-X Quick Edition IP Telephones use broadcast messages to establish association with other Avaya one-X Quick Edition peers to form a one-X Quick Edition network. The Avaya one-X Quick Edition network uses a service provider configuration to establish SIP trunking with the SES Server. SIP requests from one-X Quick Edition IP Telephones, including registration, are authenticated with a password configured on the SES Server for the corresponding SES user. Standard features for Avaya one-X Quick Edition IP Telephones, including automatic SIP registration and recovery capabilities, are retained when placed in a service provider configuration. Avaya Communication Manager is administered with SIP trunking to the SES Server for routing calls destined to the one-X Quick Edition network. When Avaya one-X Quick Edition IP Telephones are provisioned without a one-X Quick Edition G10 PSTN Gateway, each telephone will register with the SES server for handling its own incoming and outgoing calls. When Avaya one-X Quick Edition IP Telephones are provisioned with a one-X Quick Edition G10 PSTN Gateway, the gateway acts as a proxy and will register with the SES Server, on behalf of each Avaya one-X Quick Edition IP Telephone, for handling incoming and outgoing calls.

The sample network environment in **Figure 1** represents a Main location with Avaya Communication Manager connected through a WAN to a Branch location with an Avaya one-X Quick Edition network. The Main location consists of S8710 Media Servers with an Avaya G650 Media Gateway and hosts an SES Server to support SIP proxy requests from registered endpoints. The Branch location was configured with three Avaya one-X Quick Edition IP Telephones and the system-wide Auto Attendant for SIP communication with the SES Server. One Avaya one-X Quick Edition IP Telephone was mapped as an individual SIP identity in the SES Server. The other two Avaya one-X Quick Edition IP Telephones were configured in a hunt group and mapped to a corresponding SIP identity in the SES Server. The global (local) extension used by the embedded Auto Attendant for the Avaya one-X Quick Edition network was also mapped to a SIP identity in the SES Server.

The network diagram in **Figure 1** illustrates the network components that were used to verify these Application Notes. Any configuration related to the underlying infrastructure and other network elements will not be covered. Also, these Application Notes provide a sample of capabilities that is featured by Avaya one-X Quick Edition and does not cover all functionality. Please see **Section 9** of these Application Notes to view additional references.

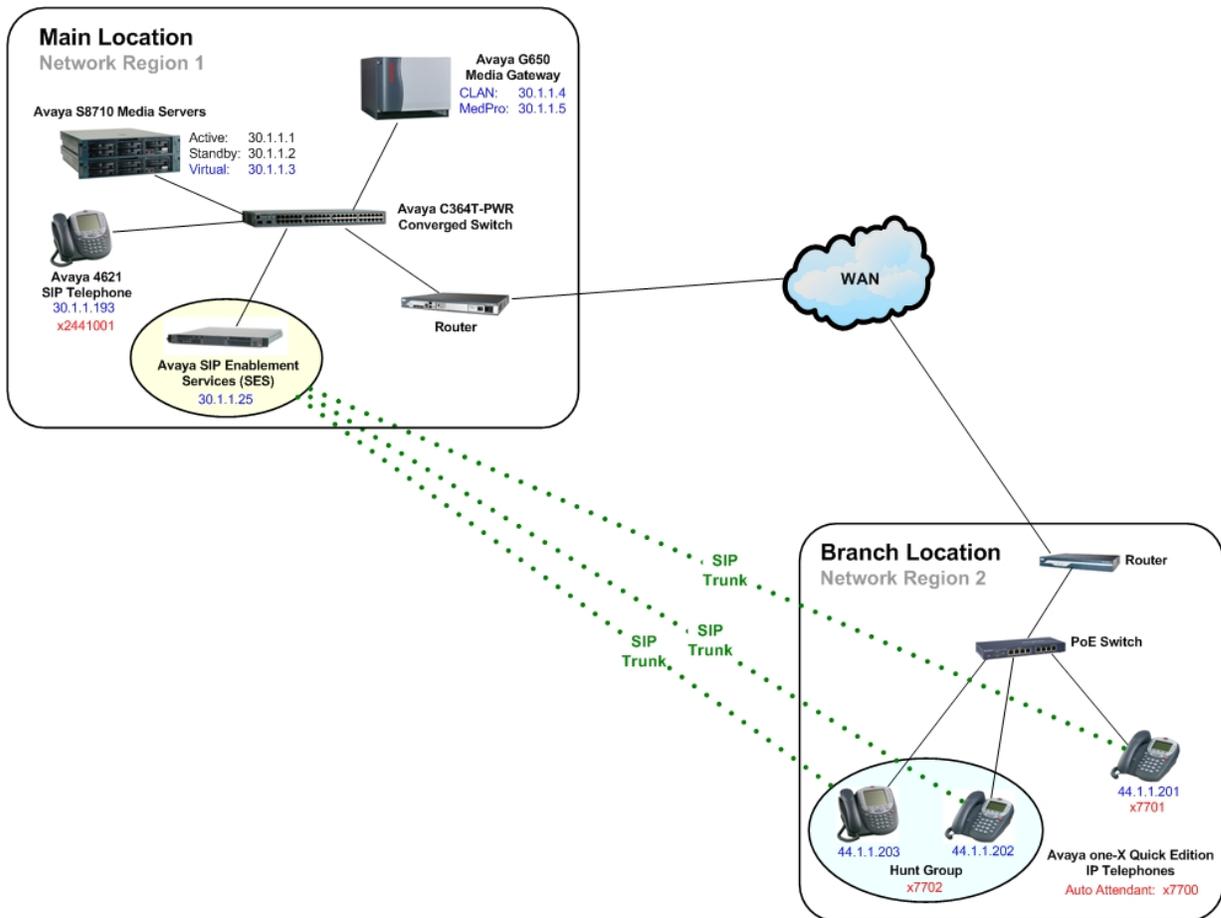


Figure 1: Network Environment

2. Equipment and Software Validated

The following table describes the equipment and software used in the above network configuration was used for validation in these Application Notes:

Equipment	Software
Avaya S8710 Media Server (2) <ul style="list-style-type: none"> Avaya Communication Manager 	3.1.2
Avaya G650 Media Gateway <ul style="list-style-type: none"> IPSI (TN2312BP) C-LAN (TN799DP) MEDPRO (TN2602AP) 	<ul style="list-style-type: none"> FW31 FW17 FW112
Avaya SIP Enablement Services	3.1.1
Avaya C363T-PWR Converged Switch	4.5.14
Avaya one-X Quick Edition IP Telephone <ul style="list-style-type: none"> Avaya 4610SW Avaya 4621SW 	3.0.0
Avaya 4621SW IP Telephone (SIP)	2.7

3. Avaya one-X Quick Edition IP Telephone Approach

Listed below are steps to manually change the network address settings for the Avaya one-X Quick Edition IP Telephone. All Avaya one-X Quick Edition IP Telephones were given a static configuration in the network environment illustrated in **Figure 1**:

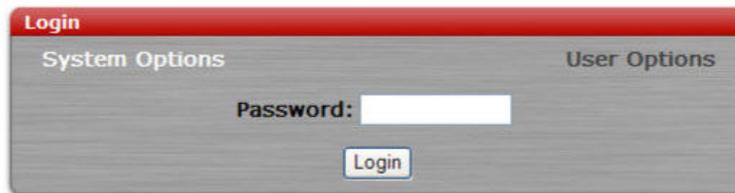
1. Upon phone startup, enter an ASCII string to create a site ID or join a created site when prompted.
2. From the main screen on the Avaya one-X Quick Edition IP Telephone, select the softkey for **System Options > Network Options**.
3. From the **Network Options** menu, select **IP Address**.
4. Select the **Chg** softkey. Assign an IP address (ex. 44.1.1.201) and select the **Next** softkey.
5. Enter the network mask (ex. 255.255.255.0) and select the **Next** softkey.
6. Enter the IP address of the default gateway and select the **Next** softkey.
7. (*Optional*) Enter the IP address of the DNS server and select the **Next** softkey
8. Select the **Save** softkey.
9. Repeat Steps 1 through 7 for other Avaya one-X Quick Edition IP Telephones.

4. Configure the Avaya one-X Quick Edition Network

Listed below are the steps used to configure the Avaya one-X Quick Edition network with standalone one-X Quick Edition IP Telephones for communication with the SES Server to enable SIP Telephony with endpoints registered to Avaya Communication Manager.

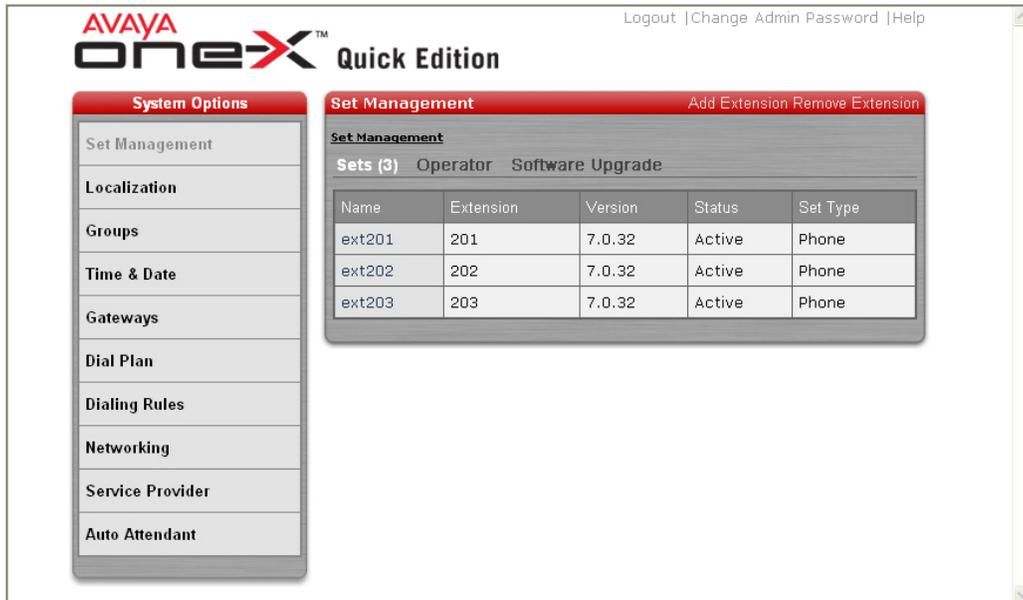
1. To access the one-X Quick Edition network, enter `http://<a.b.c.d>` at a web browser where a.b.c.d is the IP address of any Avaya one-X Quick Edition IP Telephone previously configured in Section 3. Enter a valid password at the **System Options** page to modify system parameters for the Avaya one-X Quick Edition network and click **Login**.

AVAYA
one-X™ Quick Edition

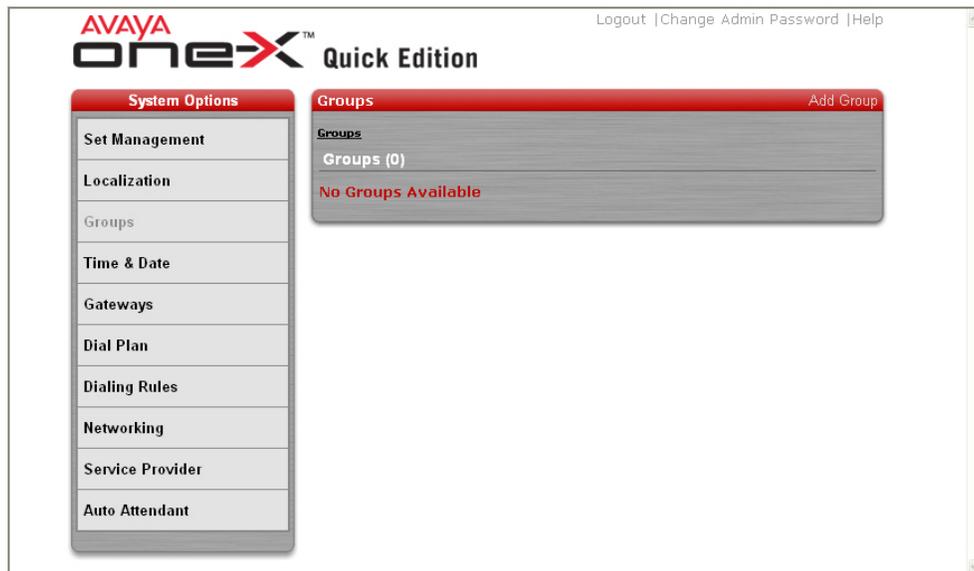


©2005 Avaya Inc. All Rights Reserved.

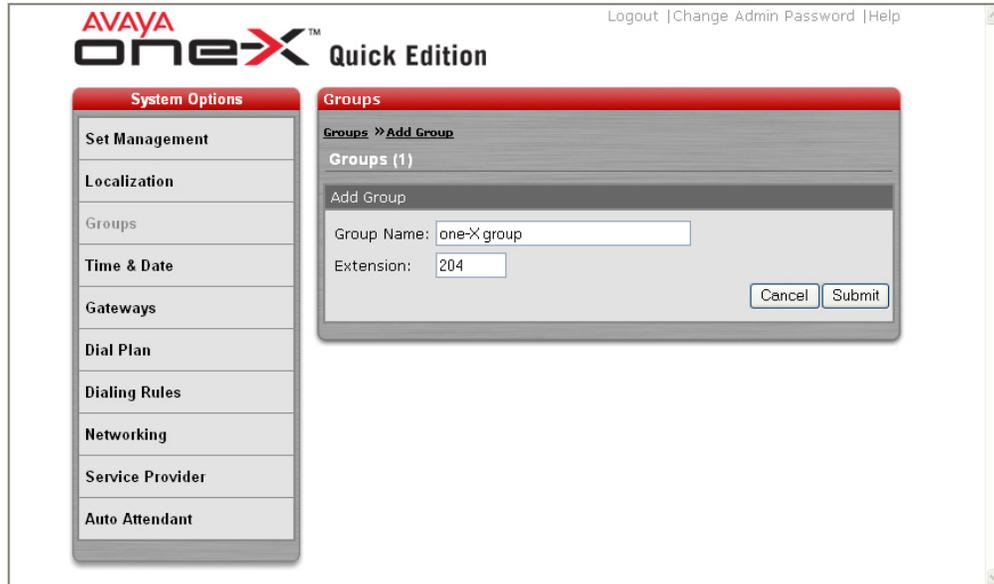
2. From the System Options menu, click the **Groups** link to open the Groups screen to configure parameters for a hunt group with Avaya one-X Quick Edition IP Telephones.



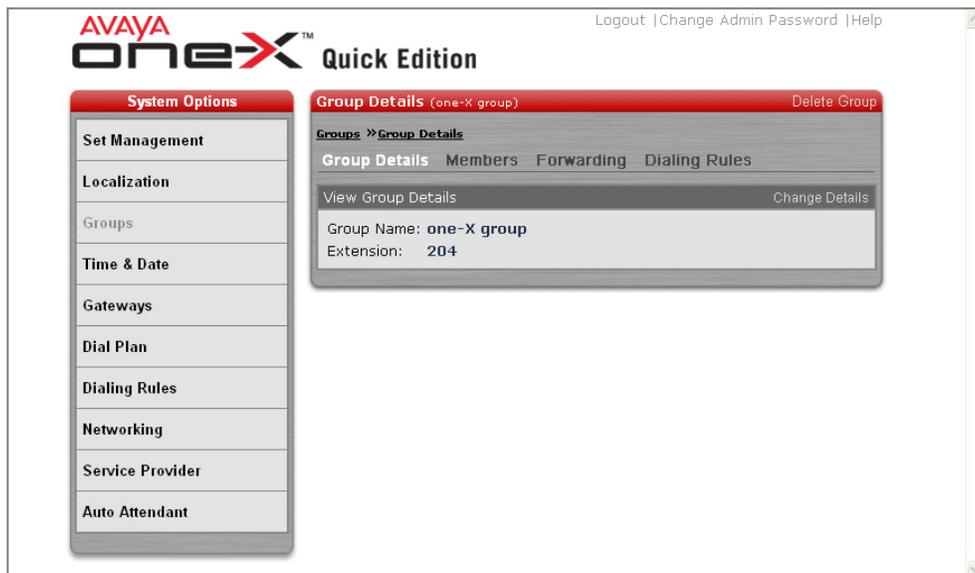
3. From the Groups screen, click the **Add Group** link to add a hunt group to the one-X Quick Edition network.



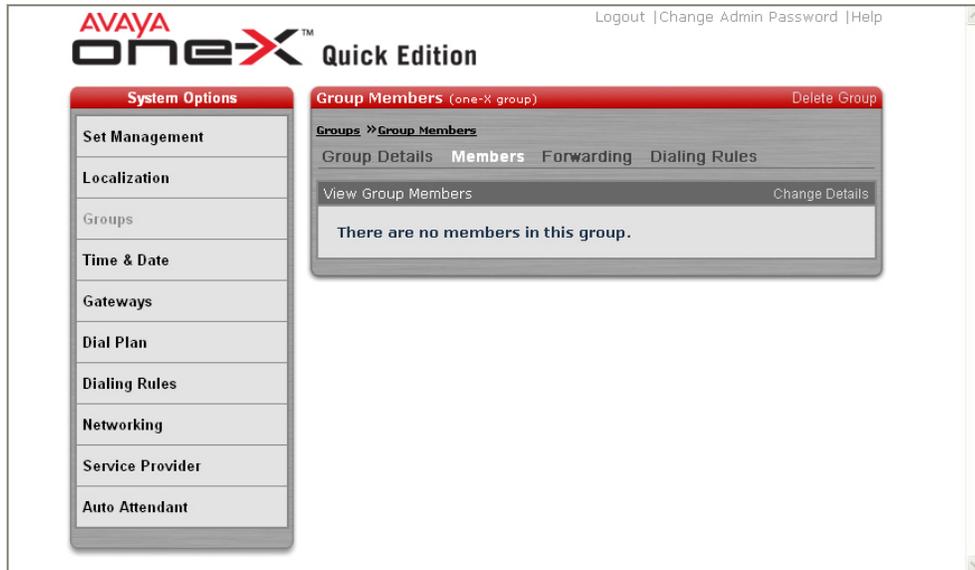
4. At the Add Group page, enter a descriptive name for the hunt group in the **Group Name** field. Enter an extension for this hunt group in the **Extension** field that will be used locally by the Avaya one-X Quick Edition network. Click the **Submit** button when finished to display the Group > Group Details page.



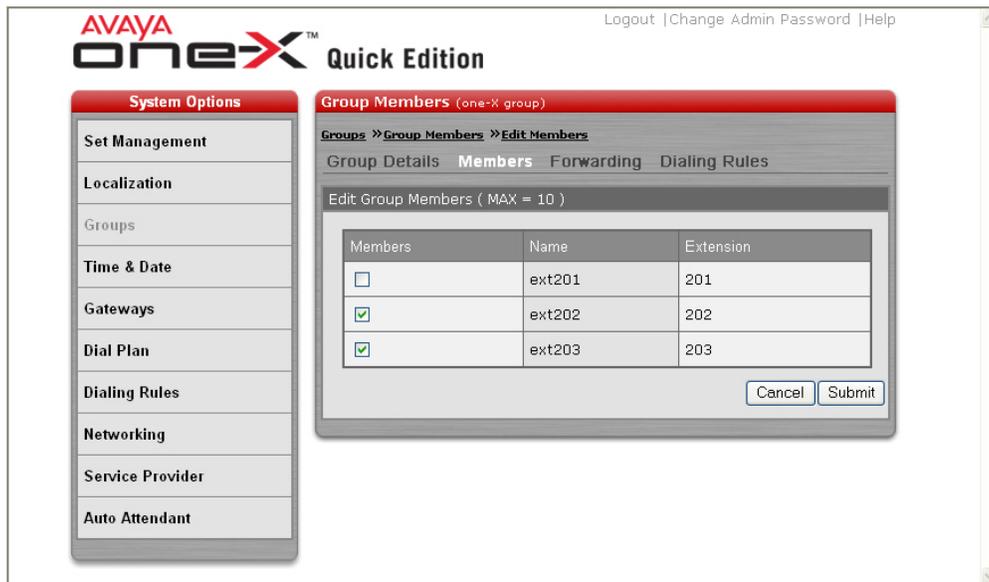
5. At the Group > Group Details page, click the **Members** link to configure membership of Avaya one-X Quick Edition IP Telephones for the hunt group.



6. At the Group > Group Members page, click the **Change Details** link to enter the Group > Group Members > Edit Members page for adding Avaya one-X Quick Edition IP Telephones to the hunt group.



7. At the Group > Group Members > Edit Members page, select the checkbox in the **Members** list for each Avaya one-X Quick Edition IP Telephone that is designated for membership to the hunt group. Click the **Submit** button when finished.



8. At the System Options menu, click the Service Provider link to add a service provider configuration to the one-X Quick Edition network. The **Service Provider > Configurations** page is displayed. Click **Add Configuration** to create a service provider profile for the one-X Quick Edition network.



9. At the Add Service Provider Configuration screen, enter a domain name associated with the SIP domain in the **Domain Name** and **Realm** fields. Enter the IP address of the SES server in the fields for **Proxy Host** and **Registrar Host**. Enter the port number to listen for SIP communication from the SES server in the fields for **Proxy Port** and **Registrar Port**.

In the **Register Expiry Time** field, enter the amount of time for the one-X Quick Edition IP Telephones to refresh registration information with the SES Server. This value is measured in seconds. Leave the remaining fields blank for **Outbound Proxy Host** and **Outbound Proxy Port**. Click the **Submit** button when finished.

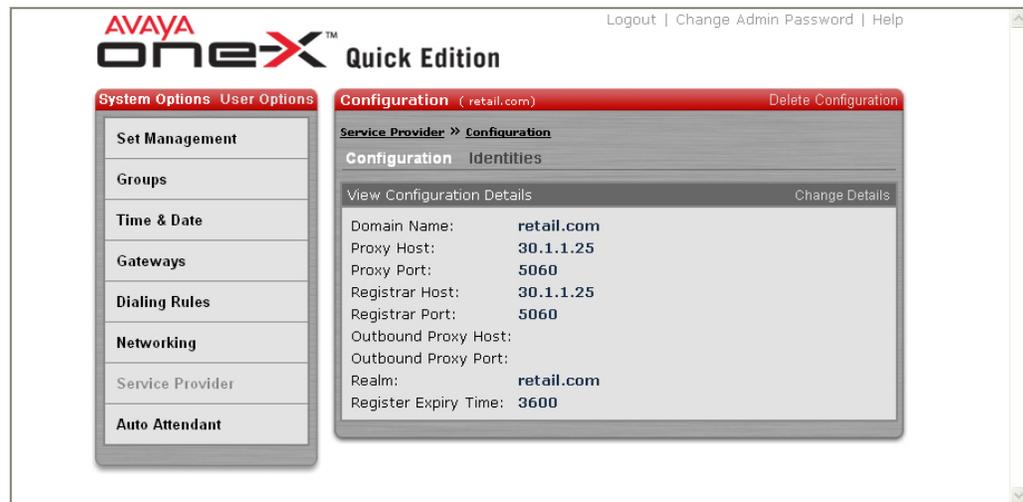
*Note: Values will be needed for **Outbound Proxy Host** and **Outbound Proxy Port** if the one-X Quick Edition network is using a Session Border Controller (SBC) for NAT filtering*

The screenshot displays the Avaya One-X Quick Edition web interface. The top navigation bar includes the Avaya One-X logo, the text 'Quick Edition', and links for 'Logout | Change Admin Password | Help'. A left-hand menu contains categories: 'System Options' and 'User Options'. Under 'System Options', the following items are listed: 'Set Management', 'Groups', 'Time & Date', 'Gateways', 'Dialing Rules', 'Networking', 'Service Provider', and 'Auto Attendant'. The 'Service Provider' category is selected, leading to a 'Service Provider' window titled 'Add Service Provider Configuration'. This window shows a list of 'Configurations (0)' and a form for adding a new configuration. The form fields are: Domain Name (retail.com), Proxy Host (30.1.1.25), Proxy Port (5060), Registrar Host (30.1.1.25), Registrar Port (5060), Outbound Proxy Host (blank), Outbound Proxy Port (blank), Realm (retail.com), and Register Expiry Time (3600). 'Cancel' and 'Submit' buttons are located at the bottom right of the form.

10. From the **System Options** menu, select **Service Provider** to display the **Service Provider > Configurations** list. From the **Service Provider > Configurations** list, click the domain name for the service provider configuration created in Steps 8 and 9.



11. At the View Configuration Details screen, click on **Identities** to enter the configuration list for one-X Quick Edition SIP identities.

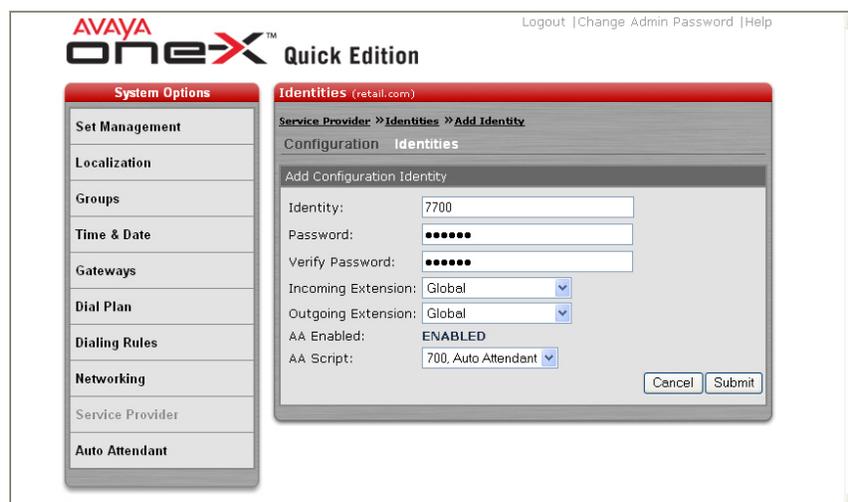


12. From the **Service Provider > Identities** screen, click **Add Identity** to add SIP identities associated with this service provider configuration for the one-X Quick Edition network. In this case, an identity is an Avaya one-X Quick Edition IP Telephone configured as a user in the service provider configuration.



13. At the Add Configuration Identity screen, enter an extension that will be assigned to the one-X Quick Edition Auto Attendant in the **Identity** field. Enter a password in the **Password** field for the one-X Quick Edition Auto Attendant extension that will be associated with the SES Server.

Select the **Global** option from the **Incoming Extension** and **Outgoing Extension** drop-down list to enable the one-X Quick Edition Auto Attendant for this identity. Select a name from the **AA Script** drop-down list to use the default Auto Attendant or a custom Auto Attendant configuration. Click **Submit** when finished.



14. At the Add Configuration Identity screen, enter an extension that will be assigned directly to a one-X Quick Edition IP Telephone in the **Identity** field. Enter a password in the **Password** field for the one-X Quick Edition IP Telephone that will be associated with the SES Server.

From the **Incoming Extension** list, select the one-X Quick Edition IP Telephone that will handle incoming calls for the specified identity. From the **Outgoing Extension** list, select the one-X Quick Edition IP Telephone that will place outgoing calls for the specified identity. Repeat this step as necessary to define the corresponding one-X Quick Edition IP Telephones provisioned as users on the SES server.

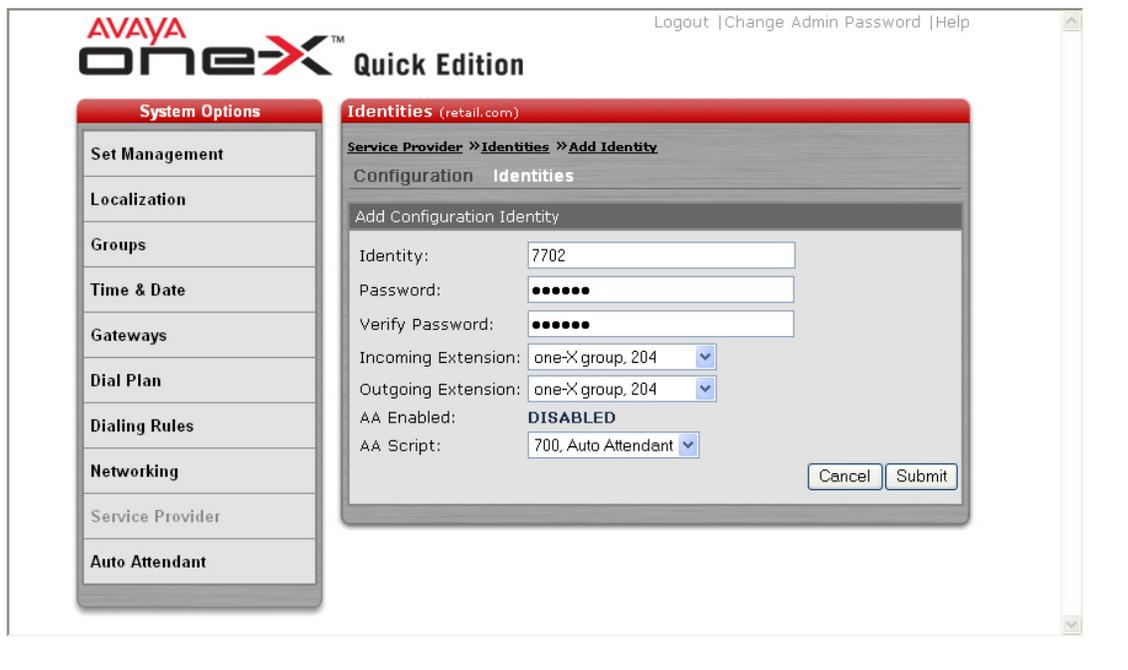
*Note: The AA Enabled field changes from **ENABLED** to **DISABLED** when a one-X Quick Edition extension is selected from the **Incoming Extension** drop-down list.*

The screenshot displays the Avaya one-X Quick Edition configuration interface. On the left is a sidebar menu with categories: System Options, Localization, Groups, Time & Date, Gateways, Dial Plan, Dialing Rules, Networking, Service Provider, and Auto Attendant. The main content area is titled 'Identities (retail.com)' and shows a breadcrumb path: 'Service Provider >> Identities >> Add Identity'. Below this is a 'Configuration Identities' section with a sub-section 'Add Configuration Identity'. The form contains the following fields: Identity (text input: 7701), Password (password input: masked with dots), Verify Password (password input: masked with dots), Incoming Extension (dropdown menu: ext201, 201), Outgoing Extension (dropdown menu: ext201, 201), AA Enabled (text field: DISABLED), and AA Script (dropdown menu: 700, Auto Attendant). 'Cancel' and 'Submit' buttons are located at the bottom right of the form.

15. At the Add Configuration Identity screen, enter an extension that will be assigned to the group of one-X Quick Edition IP Telephones created in Steps 3 through 7 in the **Identity** field. Enter a password in the **Password** field that will be used to authenticate the group of one-X Quick Edition IP Telephones with the SES Server.

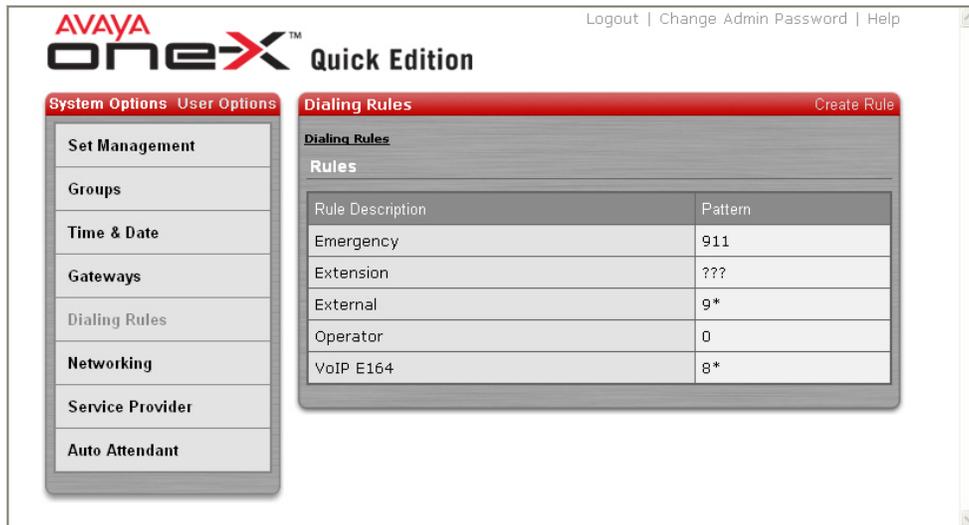
From the **Incoming Extension** list, select the group of one-X Quick Edition IP Telephones that will handle incoming calls for the specified identity. From the **Outgoing Extension** list, select the group configured in Steps 4 through 7 to enable outgoing calls from the one-X Quick Edition IP Telephones with group membership for the specified identity.

*Note: The AA Enabled field changes from **ENABLED** to **DISABLED** when a one-X Quick Edition extension is selected from the **Incoming Extension** drop-down list.*



16. From the **System Options** menu, select **Dialing Rules** to display the **Dialing Rules > Rules** list. The default **VoIP E164** Rule Description with an **8*** dialing pattern seizes the SIP trunk to the SES Server for redirecting calls from the one-X Quick Edition network to Avaya Communication Manager.

*Note: The dialing patterns for **Emergency**, **Extension**, **External**, **Operator**, and **VoIP E164** are pre-configured values and cannot be deleted or modified.*



5. Configure Avaya Communication Manager

Listed below are the steps used to configure the Avaya Communication Manager on the S8710 Media Server for integration with Avaya SIP Enablement Services and Avaya one-X Quick Edition. Start a SAT terminal session to Avaya Communication Manager and access the system using valid login credentials. These Application Notes assume the proper licensing and customer options for Avaya Communication Manager have been installed and enabled for SIP integration with Avaya SIP Enablement Services.

1. Enter the **change node-names ip** command at the SAT terminal to configure association between a node name and the IP address of the SES Server. Enter a node name under the field for **Name** and the IP address of the SES Server under the field for **IP Address**. Identify node name and IP address of the C-LAN communicating with Avaya Communication Manager.

```
change node-names ip                                     Page 1 of 1
Name                                                    IP NODE NAMES
Name                                                    IP Address
C-LAN          30 .1 .1 .4
G700-HQ1        40 .1 .1 .1
HQ-VAL          30 .1 .1 .31
MM-MAS          30 .1 .1 .9
MediaResource   30 .1 .1 .32
Medpro          30 .1 .1 .5
default         0 .0 .0 .0
exchange-mas    30 .1 .1 .19
procr           . . .
hq-ses        30 .1 .1 .25
. . .
. . .
. . .
```

2. Enter the **change dialplan analysis** command at the SAT terminal to modify the dial plan analysis table. Enter values under **Dialed String, Total Length** and set the **Call Type** field to **dac** to create a dial access code that will be used for the trunk group to SIP Enablement Services.

Enter values under **Dialed String, Total Length** and set the **Call Type** field to **ext** to create a dialing string that will be used by Avaya Communication Manager for SIP endpoints. Enter values under **Dialed String, Total Length** and set the **Call Type** field to **ext** to create a dialing string that represents SIP identities associated with the one-X Quick Edition network. This is necessary for digit analysis of extensions configured on the SES Server for the one-X Quick Edition network.

```
change dialplan analysis                                     Page 1 of 12
```

DIAL PLAN ANALYSIS TABLE								
Percent Full: 2								
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
1	3	dac	73	4	ext			
2	7	ext	74	4	ext			
2000	5	ext	77	4	ext			
244	7	ext	8	1	fac			
255	7	ext	9	1	dac			
2800	5	ext	*	3	fac			
3	3	ext	#	3	fac			
333	7	ext						
422	5	ext						
66	5	ext						
70	4	ext						

3. Enter the **change ip-codec-set <codec set number>** command at the SAT terminal to configure codec parameters for communication between Avaya Communication Manager and the Avaya one-X Quick Edition network. Enter **G.729A** in addition to the default G.711mu codec under the **Audio Codec** field. Auto Attendant and voice mail on the Avaya one-X Quick Edition network requires the G.729a codec for interoperability with Avaya Communication Manager.

Note: Avaya one-X Quick Edition IP Telephones will automatically negotiate between G.711mu and G.729a if both are provisioned in the codec configuration for Avaya Communication Manager.

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

                               IP Codec Set

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression   Per Pkt    Size(ms)
1: G.711MU      n           2          20
2: G.729A      n           2          20
3:
4:
5:
6:
7:

Media Encryption
1: none
2:
3:
```

4. Enter the **change ip-network-region <network region number>** command to define network properties used for communicating with the Avaya one-X Quick Edition network within the administrative domain. Enter the domain name used for this network region in the **Authoritative Domain** field. Enter the ip-codec set configured in the previous step in the **Codec Set** field. Navigate to Page 3 of this network region when finished.

```
change ip-network-region 2                                     Page 1 of 19
                                                           IP NETWORK REGION
  Region: 2
Location:      Authoritative Domain: retail.com
  Name:QE-Region
MEDIA PARAMETERS                                         Intra-region IP-IP Direct Audio: yes
  Codec Set: 1                                         Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048                                     IP Audio Hairpinning? n
  UDP Port Max: 3327
DIFFSERV/TOS PARAMETERS                                 RTCP Reporting Enabled? y
  Call Control PHB Value: 46                             RTCP MONITOR SERVER PARAMETERS
  Audio PHB Value: 46                                   Use Default Server Parameters? y
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5                             AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
  Idle Traffic Interval (sec): 20
  Keep-Alive Interval (sec): 5
  Keep-Alive Count: 5
```

5. On Page 3 of the **change ip-network-region <network region number>** command, configure the network region to specify call admission control between Avaya Communication Manager and the network region designated for the Avaya one-X Quick Edition Network.

In the example below, calls sourced from the one-X Quick Edition network in network region 2 are connected to the Avaya Communication Manager through a direct WAN link to network region 1. The Avaya one-X Quick Edition network is configured to use the ip-codec set administered in Step 3 and is allocated a maximum of four (4) calls when network region 2 passes call control to network region 1. Leave the remaining parameters at the default setting and submit this command for the administered network region.

```
change ip-network-region 2 Page 3 of 19
```

Inter Network Region Connection Management

src rgn	dst rgn	codec set	direct WAN	Total WAN-BW-limits	Video Norm Prio	Shr	Intervening-regions	Dyn CAC	IGAR
2	1	1	y	4 :Calls		y			n
2	2	1							
2	3								
2	4								
2	5								
2	6								
2	7								
2	8								
2	9								
2	10								
2	11								
2	12								
2	13								
2	14								
2	15								

6. Enter the **change ip-network-region <network region number>** command to define network properties used for communicating with the SES Server network within the administrative domain. Enter the domain name used for this network region in the **Authoritative Domain** field. Enter the ip-codec set configured in Step 3 in the **Codec Set** field. Navigate to Page 3 of this network region when finished.

```
change ip-network-region 1                               Page 1 of 19
                                     IP NETWORK REGION
  Region: 1
  Location:      Authoritative Domain: retail.com
    Name:SES-Region
  MEDIA PARAMETERS                                     Intra-region IP-IP Direct Audio: yes
    Codec Set: 1                                       Inter-region IP-IP Direct Audio: yes
    UDP Port Min: 5000                                     IP Audio Hairpinning? y
    UDP Port Max: 5999
  DIFFSERV/TOS PARAMETERS                               RTCP Reporting Enabled? y
    Call Control PHB Value: 46                             RTCP MONITOR SERVER PARAMETERS
    Audio PHB Value: 46                                   Use Default Server Parameters? y
    Video PHB Value: 26
  802.1P/Q PARAMETERS
    Call Control 802.1p Priority: 6
    Audio 802.1p Priority: 6
    Video 802.1p Priority: 5
  H.323 IP ENDPOINTS                                   AUDIO RESOURCE RESERVATION PARAMETERS
    H.323 Link Bounce Recovery? y                           RSVP Enabled? n
    Idle Traffic Interval (sec): 20
    Keep-Alive Interval (sec): 5
    Keep-Alive Count: 5
```

7. On Page 3 of the **change ip-network-region <network region number>** command, configure the network region to specify call admission control between Avaya Communication Manager and the network region designated for the SES Server.

In the example below, calls sourced from SIP-enabled endpoints registered to Avaya Communication Manager in network region 1 are connected to the Avaya one-X Quick Edition network through a direct WAN link to network region 2. The SIP-enabled endpoints registered to Avaya Communication Manager are configured to use the ip-codec set administered in Step 3 and is allocated unlimited calling privileges when network region 2 passes call control to network region 1. Leave the remaining parameters at the default setting and submit this command for the administered network region.

```
change ip-network-region 2 Page 3 of 19
```

Inter Network Region Connection Management

src rgn	dst rgn	codec set	direct WAN	Total WAN-BW-limits	Video Norm Prio	Shr Intervening-regions	Dyn CAC	IGAR
1	1	1						
1	2	1	y	:NoLimit		y		n
1	3	1	y	:NoLimit		y		n
1	4	1	y	:NoLimit		y		n
1	5							
1	6							
1	7							
1	8							
1	9							
1	10							
1	11							
1	12							
1	13							
1	14							
1	15							

8. Enter the **change ip-network-map** command to define address mapping to the corresponding network region for SIP-enabled endpoints registered to Avaya Communication Manager and the Avaya one-X Quick Edition network.

Under the **From IP Address** and (**To IP Address or Subnet Mask**) fields, enter the IP address range used by the Avaya one-X Quick Edition network and enter the network region configured in Steps 4 and 5 under the **Region** field. Under the **From IP Address** and (**To IP Address or Subnet Mask**) fields, enter the IP address range used by SIP-enabled endpoints registered to Avaya Communication Manager and enter the network region configured in Steps 6 and 7 under the **Region** field.

```
change ip-network-map
```

Page 1 of 32

IP ADDRESS MAPPING										
From IP Address		(To IP Address			Subnet		Region	VLAN	Emergency	
		or Mask)							Location	
									Extension	
30	.1	.1	.100	30	.1	.1	.200	1	n	
40	.1	.1	.1	40	.1	.1	.254	3	n	
44	.1	.1	.100	44	.1	.1	.200	2	n	
.		n	
.		n	
.		n	
.		n	

9. Enter the **add signaling-group <group number>** command at the SAT terminal to configure signaling parameters between Avaya Communication Manager and SES Server. Select **sip** as the signaling group type in the **Group Type** field and enter **tls** in the **Transport Method** field.

Enter the node name defined in Step 1 for the C-LAN in the **Near-end Node Name** field and enter a value for the **Near-end Listen Port**. Enter the node name defined in Step 1 for the SES Server in the **Far-end Node Name** field and enter the **Far-end Listen Port** for signaling termination.

Set a network region in the **Far-end Network Region** field that will be used by SIP Enablement Services. Enter the domain name in the **Far-end Domain** field that was defined for the network region configured in Step 6 for the SES Server. Set the field for **Direct IP-IP Audio Connections** to **n** to disable shuffling during call signaling.

```
add signaling-group 5                                     Page 1 of 1
                                                    SIGNALING GROUP
Group Number: 5                Group Type: sip
                               Transport Method: tls

Near-end Node Name: C-LAN      Far-end Node Name: hq-ses
Near-end Listen Port: 5061     Far-end Listen Port: 5061
                               Far-end Network Region: 1
Far-end Domain: retail.com

                               Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload     Direct IP-IP Audio Connections? n
                               IP Audio Hairpinning? y
Session Establishment Timer(min): 120
```

10. Enter the **add trunk-group <trunk group number>** command at the SAT terminal and select **sip** in the **Group Type** field. Enter a descriptive name for the **Group Name** field and a value for the **TAC** field that is consistent with the dial access code defined in Step 2 of this section.

Set the **Service Type** field to **tie** and under the **Signaling Group** field, enter the signaling group configured in the previous step. Enter the trunk group members that are designated for the SIP trunk in the **Number of Members** field. Each trunk group member is dynamically assigned by the trunk group.

```

add trunk-group 5                                     Page 1 of 21
                                     TRUNK GROUP

Group Number: 5                                     Group Type: sip          CDR Reports: y
  Group Name: To-SES                                COR: 1                  TN: 1          TAC: 105
  Direction: two-way                               Outgoing Display? y
  Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                                   Auth Code? n

                                               Signaling Group: 5
                                               Number of Members: 23
  
```

11. Enter the **change uniform-dialplan <dial string >** command at the SAT terminal to assign uniform dialing between the one-X Quick Edition network and Avaya Communication Manager. Enter a value under the **Matching Pattern**, **Len**, and **Del** fields for the dialing patterns designated for the one-X Quick Edition network and for Avaya Communication Manager SIP extensions.

For each dialing pattern, select **aar** in the **Net** field to send the dialed strings for AAR analysis. These values should be consistent with the dialing pattern configured in Step 2 of this section.

```

change uniform-dialplan 2                             Page 1 of 2
                                     UNIFORM DIAL PLAN TABLE
                                               Percent Full: 0

Matching      Insert      Node      Matching      Insert      Node
Pattern  Len Del  Digits  Net  Conv  Num   Pattern  Len Del  Digits  Net  Conv  Num
244      7  0    aar  n    n
3          7  0    aar  n    n
45         4  0    aar  n    n
5          5  0    aar  n    n
66         5  0    aar  n    n
77      4  0    aar  n    n
  
```

12. Enter the **change aar analysis <dial string >** command at the SAT terminal to assign a route pattern for the one-X Quick Edition network based on digit analysis. Enter values under the **Dialed String**, **Total Min** and **Total Max** fields for the one-X Quick Edition network and Avaya Communication Manager SIP extensions. These values should be consistent with the dialing pattern configured in Step 2 of this section.

Add a value under the **Route Pattern** field to reference a route pattern to the SES Server for the one-X Quick Edition network and for Avaya Communication Manager SIP endpoints. Select **aar** in the **Call Type** field for each dialed string.

```
change aar analysis 2
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE

Percent Full: 2

	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
	244	7	7	5	aar		n
	333	7	7	2	aar		n
	42	5	5	5	aar		n
	45	4	4	2	aar		n
	50	5	5	11	aar		n
	66	5	5	3	aar		n
	77	4	4	5	aar		n

13. Enter the **change route-pattern <route-pattern number>** command at the SAT terminal to administer a route pattern for the trunk group used for SIP communication with the SES Server. Under the first entry for the **Grp No** field, enter the trunk group created in Step 10 of this section. Set the **FRL** field to **0**.

```
change route-pattern 5
```

Page 1 of 3

Pattern Number: 5 Pattern Name: SIP

SCCAN? n Secure SIP? n

Grp No	FRL	NPA	Pfx Mrk	Hop Lmt	Toll List	No. Del	Inserted Digits	DCS/ QSIG Intw	IXC
1:	5	0						n	user
2:								n	user
3:								n	user
4:								n	user
5:								n	user
6:								n	user

	BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No. Dgts	Numbering Format	LAR
	0	1	2	3	4	W	Request				
1:	y	y	y	y	y	n	n			rest	none
2:	y	y	y	y	y	n	n			rest	none
3:	y	y	y	y	y	n	n			rest	none
4:	y	y	y	y	y	n	n			rest	none
5:	y	y	y	y	y	n	n			rest	none

14. Enter the **add station <extension>** command at the SAT terminal to create a SIP extension for Avaya Communication Manager. The extension must be consistent with the dialing plan configured in Step 2 of this section.

Set the **Type** field to **6408D+** and set the **Port** field to **X**. Enter a password with at least six digits for the extension in the **Security Code** field. Repeat this step as necessary for other SIP-enabled extensions on Avaya Communication Manager.

```
add station 2441001                                     Page 1 of 5
                                                    STATION
Extension: 2441001                                Lock Messages? n          BCC: 0
Type: 6408D+                                     Security Code: ***** TN: 1
Port: X                                          Coverage Path 1:         COR: 1
Name: sip-hq1                                       Coverage Path 2:         COS: 1
                                                    Hunt-to Station:

STATION OPTIONS
    Loss Group: 2                                     Personalized Ringing Pattern: 1
    Data Module? n                                   Message Lamp Ext: 2441001
    Speakerphone: 2-way                             Mute Button Enabled? y
    Display Language: english

                                                    Media Complex Ext:
                                                    IP SoftPhone? n
```

15. Enter the **add off-pbx-telephone station-mapping** command at the SAT terminal to map Avaya Communication Manager SIP extensions with extensions configured on the SES Server. Add the extension administered in the previous step under the fields for **Station Extension** and **Phone Number**. Select **OPS** under the **Application** field to assign this extension as a SIP-enabled endpoint.

Enter the trunk group number configured in Step 10 of this section under the **Trunk Selection** field. Repeat this step as necessary for other extensions on Avaya Communication Manager using the SES Server.

```
add off-pbx-telephone station-mapping                 Page 1 of 2
                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

Station      Application  Dial   Phone Number  Trunk   Configuration
Extension
2441001      OPS         -     2441001     5
-
-
-
-
```

16. Enter the **change public-unknown-numbering <extension digits>** command at the SAT terminal to expose the numbering information between Avaya Communication Manager SIP extensions and the one-X Quick Edition network during call signaling.

Enter values under the **Total Ext Length**, **Total Ext Code** and **CPN Len** fields with extension digits for Avaya Communication Manager SIP extensions. Repeat with extension digits for the one-X Quick Edition network communicating with the SES Server. These values must be consistent with the dialing plan configured in Step 2 of this section.

```
change public-unknown-numbering 4 Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT
Total
Total
Ext Ext Trk CPN CPN Ext Ext Trk CPN CPN
Len Code Grp(s) Prefix Len Len Code Grp(s) Prefix Len
4 45 5 50 7 244 7 333 5 42
```

6. Configure Avaya SIP Enablement Services

Listed below are the steps used to configure Avaya SIP Enablement Services as home/edge SIP proxy for integration with Avaya Communication Manager and Avaya one-X Quick Edition. These Application Notes assume the proper licensing and initial setup for the SES Server has been provisioned.

1. Using a web browser, enter `http://<a.b.c.d>` at a web browser where a.b.c.d is the IP address of the SES server. At the prompt, enter valid login credentials in the **Logon ID** and **Password** fields. Click **Logon** when finished to display the administration and maintenance web interface.



The screenshot shows a web browser window displaying the Avaya Integrated Management Standard Management Solutions interface. The page features the Avaya logo in the top left corner and the text "Integrated Management Standard Management Solutions" in the top right. A "Help" link is visible in the top left. The main content area contains a "Logon" form with two input fields: "Logon ID" (containing the text "admin") and "Password" (containing a series of dots). A "Logon" button is positioned below the password field. At the bottom of the page, there is a copyright notice: "© 2006 Avaya Inc. All rights reserved."

2. Click **Launch Administration Web Interface** to open the SIP Server Management window containing the options for SES Server administration.



3. At the SIP Server Management window, click on **System Properties** under the **Server Configuration** menu to view the SES configuration. If it is not populated, enter the domain name used by the SIP environment in the **SIP Domain** field. Leave the remaining fields at default settings and click the **Update** button.

AVAYA Integrated Management
SIP Server Management
Server: 30.1.1.25

Help Exit

Top

- Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
- Adjunct Systems
- Services
- Server Configuration
 - System Properties**
 - Admin Accounts
 - License
 - IM Log Settings
 - SNMP Configuration
- Certificate Management
- IM Logs
- Trace Logger
- Export/Import to ProVision

Edit System Properties

SES Version SES-3.1.1.0-114.0
System Configuration simplex
Host Type home/edge

SIP Domain*

Note that the DNS domain is: retail.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

License Host*

Network Properties

Local IP 30.1.1.25
Local Name hq-ses.retail.com
Logical IP 30.1.1.25
Logical Name hq-ses.retail.com
Gateway IP Address 30.1.1.254

Redundant Properties

Management Device SAMP

Fields marked * are required.

Update

4. At the SIP Server Management window, click on **Add** under the **Media Servers** menu to configure SIP trunking between the SES Server and Avaya Communication Manager. Enter a name for the endpoint terminating the SIP trunk under **Media Server Interface Name**. Select the SES Server used in this configuration from the **Host** drop-down list.

In the **SIP Trunk Link Type** field, select the dialog box for **TLS** to define the transport method. Enter the IP address of the terminating SIP endpoint in the **SIP Trunk IP Address** field. Leave the remaining fields at default settings and click the **Add** button.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server IP '30.1.1.25'. A navigation menu on the left lists various system components, with 'Media Servers' and 'Add' highlighted. The main content area is titled 'Add Media Server Interface' and contains the following fields:

- Media Server Interface Name***: A text input field containing 'C-LAN'.
- Host**: A dropdown menu showing '30.1.1.25'.
- SIP Trunk Link Type**: Radio buttons for 'TCP' and 'TLS', with 'TLS' selected.
- SIP Trunk IP Address***: A text input field containing '30.1.1.4'.
- Media Server** section with input fields for:
 - Media Server Admin Address (see Help)
 - Media Server Admin Login
 - Media Server Admin Password
 - Media Server Admin Password Confirm

Fields marked * are required. An 'Add' button is located at the bottom left of the form area.

5. At the SIP Server Management window, click on **List** under the **Media Servers** menu to display the media server interface configured in the previous step. Click on **Map** under **Commands** to list map statements for the configured media server interface.



6. At the List Media Server Address Map window, click on **Add Another Contact** to create a media server contact for redirecting calls to Avaya Communication Manager.



7. At the Add Media Server Contact window, enter the string used for routing traffic to Avaya Communication Manager in the **Contact** field. Click the **Add** button when finished to return to the List Media Server Address Map window.

Use the Linux regular expressions below to configure the map string for the media server contact.

sip: Indicates the protocol used
\$(user) Variable for user portion of the SIP message
@ x.x.x.x Format for Media Server IP address (ex. C-LAN)
5061: Port number used by TLS transport method.
transport = tls Indicates transport method.



8. At the List Media Server Address Map window, click on **Add Another Map** to create an address map statement for the media server contact configured in the previous step.



9. At the Add Media Server Address Map window, enter a descriptive name for identifying the address map statement in the **Name** field. Enter the map string used to match incoming digits for Avaya Communication Manager in the **Pattern** field. The values for the matching pattern should be consistent with the dial plan configured on Avaya Communication Manager.

Place a check mark in the **Replace URI** box to indicate this pattern will be forwarded by the **Host** shown. Click the **Add** button when finished to return to the List Media Server Address Map window.

Use the Linux regular expressions below to configure the map string for the media server contact:

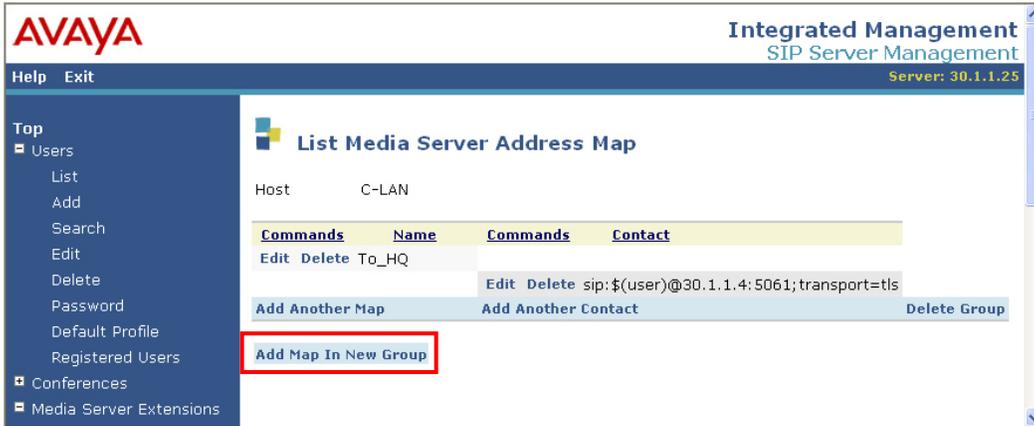
- ^ Matches first line in initial SIP message
- sip:** Indicates the protocol used
- 0-9** Match a specific digit
- [0-9]** Match any digits
- *** Indicates any digit and length

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The title bar includes the Avaya logo, the text 'Integrated Management SIP Server Management', and the server IP 'Server: 30.1.1.25'. A navigation menu on the left lists various system components. The main content area is titled 'Add Media Server Address Map' and contains the following form fields:

- Host: C-LAN
- Name*: To_HQ
- Pattern*: ^sip:2[0-9]*
- Replace URI:

Below the form fields is an 'Add' button and a note: 'Fields marked * are required.'

10. At the List Media Server Address Map window, click on **Add Map In New Group** to create an address map statement in a new group.



11. At the Add Media Server Address Map window, enter a descriptive name for identifying the address map statement in the **Name** field. Enter the map string used to match incoming digits for the Avaya one-X Quick Edition network in the **Pattern** field.

Place a check mark in the **Replace URI** box to indicate this pattern will be forwarded by the **Host** shown. Click the **Add** button when finished to return to the List Media Server Address Map window.

Use the Linux regular expressions below to configure the map string for the media server contact.

^	Matches first line in initial SIP message
sip:	Indicates the protocol used
0-9	Match a specific digit
[0-9]	Match any digits
*	Indicates any digit and length

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The title bar includes the Avaya logo, the text 'Integrated Management SIP Server Management', and 'Server: ses-hq'. A navigation menu on the left lists various system components. The main content area is titled 'Add Media Server Address Map' and contains the following fields:

- Host: C-LAN
- Name*: To_Quick_Edition
- Pattern*: ^sip:77[0-9]*
- Replace URI:

Below the fields, it states 'Fields marked * are required.' and there is an 'Add' button.

12. At the List Media Server Address Map window, click on **Edit** under **Commands** to modify the media server contact in the new group created in the previous step.



13. At the Edit Media Server Contact window, modify the string in the **Contact** field to enable routing between the SES Server and the one-X Quick Edition network. Click the **Update** button when finished to return to the List Media Server Address Map window.

Use the Linux regular expressions below to configure the map string for the media server contact.

- sip:** Indicates the protocol used
- \$(user)** Variable for user portion of the SIP message
- @ x.x.x.x** Format for SES Server IP address
- 5061:** Port number used by TLS transport method.
- transport = tls** Indicates transport method.



14. At the SIP Server Management window, click on **Add** under the **Users** menu to configure a SIP identity for an endpoint registered to Avaya Communication Manager. Enter an extension for the user in the **Primary Handle** field. This value should be consistent with the dial plan configured on Avaya Communication Manager.

Enter a password for the user in the fields for **Password** and **Confirm Password**. Select the SES Server managing the SIP Domain from the **Host** drop-down list. Enter a descriptive name for the user in the fields for **First Name** and **Last Name**. Select the check box for **Add Media Server Extension** and click the **Add** button when finished to display the Add Media Server Extension screen.

AVAYA Integrated Management
SIP Server Management
Server: 30.1.1.25

Help Exit

Top

- Users
 - List
 - Add
 - Search
 - Edit
 - Delete
 - Password
 - Default Profile
 - Registered Users
- Conferences
- Media Server Extensions
 - Emergency Contacts
- Hosts
- Media Servers
 - List
 - Add
- Adjunct Systems
 - Services
- Server Configuration
- Certificate Management
- IM Logs
- Trace Logger
- Export/Import to ProVision

Add User

Primary Handle* 2441001

User ID 2441001

Password* ●●●●●●

Confirm Password* ●●●●●●

Host* 30.1.1.25

First Name* sip

Last Name* user1

Address 1

Address 2

Office

City

State

Country

Zip

Add Media Server Extension

Fields marked * are required.

Add

15. At the Add Media Server Extension screen, enter the extension configured as an off-pbx-telephone station in Avaya Communication Manager in the **Extension** field. Select the name of the media server interface from the **Media Server** drop-down list. Click the **Add** button when finished.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top left corner features the Avaya logo. The top right corner shows the title 'Integrated Management SIP Server Management' and the server IP address 'Server: 30.1.1.25'. A navigation menu on the left includes 'Help', 'Exit', and a 'Top' section with options like 'Users', 'List', 'Add', 'Search', 'Edit', 'Delete', 'Password', 'Default Profile', 'Registered Users', 'Conferences', and 'Media Server Extensions'. The main content area is titled 'Add Media Server Extension' and contains the following text: 'Add Media Server extension for user 2441001.' Below this, there are two input fields: 'Extension*' with the value '2441001' and 'Media Server' with a dropdown menu set to 'C-LAN'. A note below the fields states 'Fields marked * are required.' and an 'Add' button is positioned at the bottom of the form.

16. At the SIP Server Management window, click on **Add** under the **Users** menu to configure a SIP identity for the one-X Quick Edition Auto Attendant. Enter the extension associated with the one-X Quick Edition Auto Attendant in the **Primary Handle** field. This value should be consistent with the identity provisioned for the one-X Quick Edition Auto Attendant in Step 13 of Section 4 as well as the dial plan configured on Avaya Communication Manager.

Enter a password for the user in the fields for **Password** and **Confirm Password**. Select the SES Server managing the SIP Domain from the **Host** drop-down list. Enter a descriptive name for the user in the fields for **First Name** and **Last Name**. Click the **Add** button when finished.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The main content area is titled "Add User" and contains a form with the following fields:

Primary Handle*	7700
User ID	7700
Password*	•••••
Confirm Password*	•••••
Host*	30.1.1.25
First Name*	one-X
Last Name*	AA
Address 1	
Address 2	
Office	
City	
State	
Country	
Zip	
Add Media Server Extension	<input type="checkbox"/>

Fields marked * are required.

The "Add" button is located at the bottom left of the form.

17. At the SIP Server Management window, click on **Add** under the **Users** menu to configure a SIP identity for a one-X Quick Edition IP Telephone. Enter the extension associated with the one-X Quick Edition IP Telephone in the **Primary Handle** field. This value should be consistent with the identity provisioned for the one-X Quick Edition IP Telephone as a direct extension in Step 14 of Section 4. This value should also be consistent with the dial plan configured on Avaya Communication Manager.

Enter a password for the user in the fields for **Password** and **Confirm Password**. Select the SES Server managing the SIP Domain from the **Host** drop-down list. Enter a descriptive name for the user in the fields for **First Name** and **Last Name**. Click the **Add** button when finished.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server name 'Server: ses-hq'. A left-hand menu lists various management options such as 'Users', 'Conferences', and 'Media Server Extensions'. The main content area is titled 'Add User' and contains a form with the following fields:

Primary Handle*	7701
User ID	7701
Password*	•••••
Confirm Password*	•••••
Host*	30.1.1.25
First Name*	one-X
Last Name*	User
Address 1	
Address 2	
Office	
City	
State	
Country	
Zip	
Add Media Server Extension	<input type="checkbox"/>

Fields marked * are required. An 'Add' button is located at the bottom of the form.

18. At the SIP Server Management window, click on **Add** under the **Users** menu to configure a SIP identity for the hunt group of Avaya one-X Quick Edition IP Telephones. Enter the extension associated with the one-X Quick Edition group in the **Primary Handle** field. This value should be consistent with the identity provisioned in Step 15 of Section 4 for the one-X Quick Edition group as well as the dial plan configured on Avaya Communication Manager.

Enter a password for the user in the fields for **Password** and **Confirm Password**. Select the SES Server managing the SIP Domain from the **Host** drop-down list. Enter a descriptive name for the user in the fields for **First Name** and **Last Name**. Click the **Add** button when finished.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes the Avaya logo, the title 'Integrated Management SIP Server Management', and the server name 'Server: ses-hq'. A left-hand menu lists various system management options, with 'Users' expanded to show 'Add User'. The 'Add User' form contains the following fields:

Primary Handle*	7702
User ID	7702
Password*	•••••
Confirm Password*	•••••
Host*	30.1.1.25
First Name*	one-X
Last Name*	Group
Address 1	
Address 2	
Office	
City	
State	
Country	
Zip	
Add Media Server Extension	<input type="checkbox"/>

Fields marked * are required. An **Add** button is located at the bottom of the form.

7. Verification

These Application Notes were confirmed using the following verification steps listed below.

1. Place call between the Avaya one-X Quick Edition IP Telephone with a direct extension and a station configured on Avaya Communication Manager. Verify voice quality is acceptable.
2. Place call from a station configured on Avaya Communication Manager to the extension configured for the hunt group with Avaya one-X Quick Edition IP Telephones on the SES Server. Verify all Avaya one-X Quick Edition IP Telephones configured in the huntgroup are ringing and voice quality is acceptable when call is answered by one of the telephones.
3. Place 4 calls between stations configured on Avaya Communication Manager and the Avaya one-X Quick Edition IP Telephones on the SES Server. Place a fifth call between stations configured on Avaya Communication Manager and the Avaya one-X Quick Edition IP Telephones on the SES Server. Verify the fifth call is never established due to call control performed by Avaya Communication Manager.
4. Place call from a station configured on Avaya Communication Manager to the extension configured for one-X Quick Edition Auto Attendant on the SES Server. Verify the one-X Quick Edition Auto Attendant answers incoming calls after specified number of rings and transfers incoming calls to the appropriate extension.
5. Disconnect the Avaya one-X Quick Edition network from the WAN. Verify calls between the Avaya one-X Quick Edition IP Telephones and Avaya Communication Manager are unsuccessful. Verify local calls between Avaya one-X Quick Edition IP Telephones are successful and voice quality is acceptable.

7.1. Avaya Communication Manager Verification

1. Enter the **status signaling-group <group number>** command at the SAT terminal in Avaya Communication Manager. Verify **in-service** status of the signaling group defined for the SES Server under the **Group State** field.

```
status signaling-group 5
                        STATUS SIGNALING GROUP

      Group ID: 5                Active NCA-TSC Count: 0
      Group Type: sip            Active CA-TSC Count: 0
      Signaling Type: facility associated signaling
      Group State: in-service
```

2. Enter the **status trunk-group <group number>** command at the SAT terminal. Verify **in-service/idle** or **in-service/active** status of the trunk group defined for the SES Server under the **Service State** field.

```
status trunk 5 Page 1
```

TRUNK GROUP STATUS

Member	Port	Service State	Mtce Connected Ports Busy
0005/001	T00119	in-service/idle	no
0005/002	T00120	in-service/idle	no
0005/003	T00121	in-service/idle	no
0005/004	T00122	in-service/idle	no
0005/005	T00123	in-service/idle	no
0005/006	T00124	in-service/idle	no
0005/007	T00125	in-service/idle	no
0005/008	T00126	in-service/idle	no
0005/009	T00127	in-service/idle	no
0005/010	T00128	in-service/idle	no
0005/011	T00129	in-service/idle	no
0005/012	T00130	in-service/idle	no
0005/013	T00131	in-service/idle	no

2. Enter the **status off-pbx-telephone station <extension>** command at the SAT terminal during an active call with a Station configured on Avaya Communication Manager. Verify status of SIP-enabled station.

```
status off-pbx-telephone station 2441001
```

OFF PBX TELEPHONE STATUS

Appl No	Type	Trk/Mem Grp	Port	Connected Ports
1	OPS	0005/001	T00119	T00136

7.2. SIP Enablement Services Verification

1. At the SIP Server Management window, click on **Registered Users** under the **Users** menu. Leave all parameters blank and select the checkbox for **Include Registered Users**. Click the **Search** button when finished to view users registered to the SES Server.

Note: Search information can also be entered for a specific extension designated for one-X Quick Edition that was provisioned on the SES Server.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes the Avaya logo, 'Help Exit', and 'Integrated Management SIP Server Management Server: 30.1.1.25'. A left-hand menu lists various management options, with 'Registered Users' selected. The main content area is titled 'Registered Users on 30.1.1.25' and contains a 'Search Registered Users' section. This section includes input fields for 'Handle', 'First Name', 'Last Name', and 'Address'. Below these fields are two checkboxes: 'Include Registered Users' (checked and highlighted with a red box) and 'Include Provisioned Users' (unchecked). A 'Search' button is located below the checkboxes. At the bottom of the page, there is a 'Task:' dropdown menu set to 'Reload-complete' and a 'Submit' button.

2. At the Registered Users screen, verify registration status for the one-X Quick Edition Auto Attendant (7700), the one-X Quick Edition IP Telephone with direct extension (7701) and the one-X Quick Edition group (7702) administered as SIP identities in the SES Server. For Auto Attendant and the one-X Quick Edition group, one of the one-X Quick Edition IP Telephones in the network will register to receive all incoming calls and outgoing calls on behalf of the other telephones.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The main content area displays 'Registered Users on ses-hq' with a table of registered contacts. The table has two columns: 'Handle and Name' and 'Address'. The first three rows are highlighted with a red box, corresponding to the entities mentioned in the text: 7700@retail.com (AA, one-X), 7701@retail.com (User, one-X), and 7702@retail.com (Group, one-X). The remaining four rows show other registered users with handles like 42200@retail.com through 42205@retail.com.

Handle and Name	Address
<input type="checkbox"/> 7700@retail.com AA, one-X	sip:7700@44.1.1.202
<input type="checkbox"/> 7701@retail.com User, one-X	sip:7701@44.1.1.202
<input type="checkbox"/> 7702@retail.com Group, one-X	sip:7702@44.1.1.201
<input type="checkbox"/> 42200@retail.com Auto Attendant, one-X	sip:42200@55.1.1.204
<input type="checkbox"/> 42203@retail.com User3, one-X	sip:42203@55.1.1.203
<input type="checkbox"/> 42204@retail.com User4, one-X	sip:42204@55.1.1.204
<input type="checkbox"/> 42205@retail.com rs5-group, one-X	sip:42205@55.1.1.204

3. At the Registered Users screen, verify registration status of the SIP-enabled endpoint (2441001) configured in Avaya Communication Manager that is administered as a SIP identity in the SES Server.



8. Conclusion

As depicted by these Application Notes, an Avaya one-X Quick Edition network can be implemented as a small office solution supporting call processing with Avaya Communication Manager through SIP Enablement Services. For implementation of one-X Quick Edition without a one-X Quick Edition G10 PSTN Gateway

For the configuration used in these Application Notes, the Avaya one-X Quick Edition network at the Branch location is dependent upon reliable WAN connectivity to the SES Server at the Main location. An Avaya one-X Quick Edition implementation without the Avaya one-X Quick Edition G10 PSTN Gateway cannot provide failover or redundancy for calls destined to Avaya Communication Manager in the event of WAN failure.

9. References

The following references can be found at <http://support.avaya.com>:

- Avaya one-X Quick Edition Administrator Guide, Release 3.0.0, November 2006
- Avaya one-X Quick Edition Telephone User Guide, Release 3.0.0, November 2006
- Installing and Administering SIP Enablement Services Release 3.1.1, December 2006
- SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server, February 2006
- Administration for Network Connectivity for Avaya Communication Manager, Issue 11, February 2006

©2007 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com