Avaya Solution & Interoperability Test Lab

Configuring a Phone Emulating an Intel PBX IP Media Gateway for use with an Avaya™ S8100 Media Server / Avaya™ G600 Media Gateway - Issue 1.0

Abstract

These Application Notes provide instructions on how to configure an Intel PBX IP Media Gateway set up in phone emulation mode for use with an Avaya S8100 Media Server / Avaya G600 Media Gateway using point-to-point routing. A sample configuration using H.323 and SIP VoIP endpoints has been provided to help demonstrate this capability.

1. Introduction

These Application Notes describe how to configure an Intel PBX IP Media Gateway (PBXMG) set up in phone emulation mode for use with an Avaya S8100 Media Server / Avaya G600 Media Gateway (S8100/G600) using point-to-point routing. The PBXMG operating in Phone Emulation mode provides telephony connections between the S8100/G600 and VoIP endpoints (H.323 or SIP) by emulating digital telephone sets to the S8100/G600. It also translates protocols, as well as, converts the media formats between the two. The PBXMG consists of 8 digital telephony ports and 1 LAN port. The digital telephony ports are connected to digital line ports on the S8100/G600 and the LAN port is connected to a network switch. In Point-to-Point Routing, each PBXMG digital telephony port is directly mapped to a single IP terminal address, i.e., VoIP endpoint.

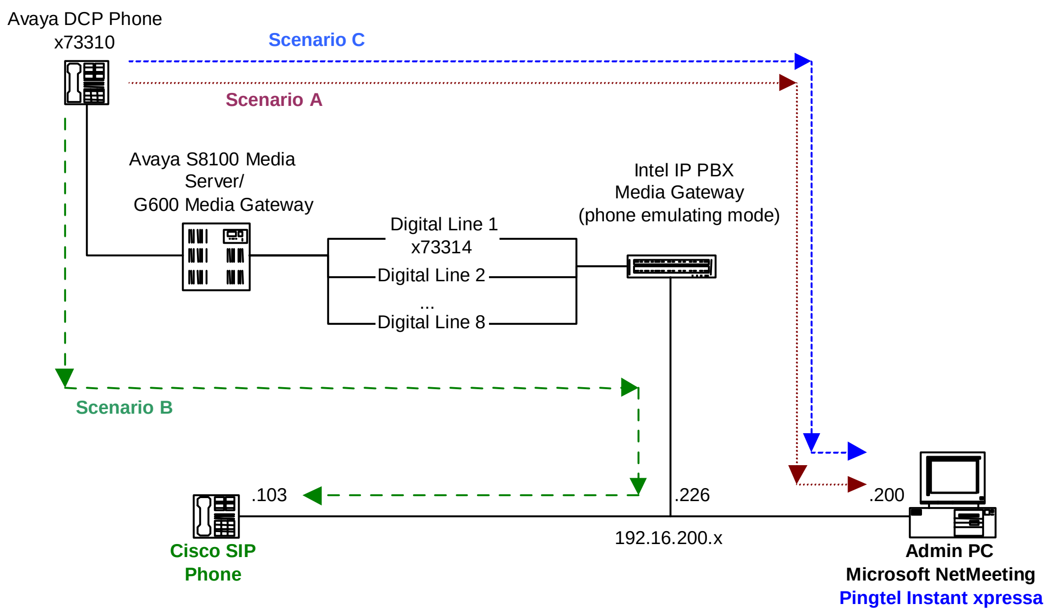

Figure 1: Sample Configuration Diagram

The configuration depicted in **Figure 1** was used to verify basic point-to-point calls using the following calling scenarios:

H.323 Emulating	Scenario A	DCP <-> S8100/G600 <-> PBXMG <-> MS NetMeeting
SIP Emulating	Scenario B	DCP <-> S8100/G600 <-> PBXMG <-> Cisco SIP IP Phone
	Scenario C	DCP <-> S8100/G600 <-> PBXMG <-> Pingtel instant xpressa

Table 1: Calling Scenarios

Note 1: These Application Notes address the Phone Emulation mode and point-to-point routing functionality of the Intel PBX IP Media Gateway without the use of a VoIP Address Translator (GateKeeper for H.323, SIP Registration Server and/or Proxy Server for SIP).

Note 2: The configuration depicted in **Figure 1** supports up to 8 digital lines to the PBX and 8 VoIP endpoints (either 8 H.323 or 8 SIP) using point-to-point routing with the exception of the Cisco SIP IP Phone. According to Intel, additional Cisco SIP IP Phones would require the use of a SIP Registration Server or Proxy Server. This exception does not apply to the Pingtel xpressa IP phones (SIP phones).

Note 3: Phone Driving mode and pooled routing are beyond the scope of this document.

Note 4: The Intel PBX IP Media Gateway supports the following codecs: G.711, G.729a, and G.723.1. The point-to-point call scenarios listed in **Table 1** were verified to work using the following codecs:

	Codecs Used	**Notes**
Scenario A (using Microsoft NetMeeting)	G.711, G.723.1	G.729a is not provided with default installation of NetMeeting.
Scenario B (using Cisco SIP 7940 IP Phone)	G.711, G.729a	
Scenario C (using Pingtel Instant xpressa)	G.711, G.729a	

Table 2: Codecs used by Calling Scenarios

2. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	**Software**
Avaya™ S8100 Media Server / Avaya™ G600 Media Gateway	R011c.02.0.110.4
Avaya™ TN2224 2-wire Digital Line Card	000005
Avaya™ 8410D Digital Telephone	-
Cisco SIP IP Phone	Application Load ID: P0S3-04-1-00 Boot Load ID: PC03A300
Intel PBX IP Media Gateway Model: PIMG80PBXDNI	DSP: 9.1 Flash App: S2_3.0.2 Flash DSP: 9.1 Boot: 4.1
Microsoft NetMeeting	3.01
Pingtel instant xpressa	2.1.8

3. Configuration

Each of the calling scenarios identified in **Table 1** required a different set of configuration steps. This was due to the following reasons:

- Two flavors of phone emulation modes: H.323 emulating and SIP emulating.
- Different endpoints used for H.323 and SIP configurations.
- Both SIP endpoints (Cisco SIP IP Phone and Pingtel xpressa) require different configurations.

Due to the different options available, the configuration instructions in the sub-sections that follow should not be followed sequentially but rather based on the desired configuration (see **Table 3**).

To configure Phone Emulating Mode:	**Using VoIP endpoint:**	**Refer to the following configuration sections:**
H.323 Emulating	MS NetMeeting	3.1 - Configure the Intel PBX IP Media Gateway 3.1.1 - Configure H.323 Emulating mode 3.2 - Configure the S8100/G600 3.3 - Configure the VoIP Endpoint(s) – MS NetMeeting
SIP Emulating	Pingtel instant xpressa	3.1 - Configure the Intel PBX IP Media Gateway 3.1.2 - Configure SIP Emulating mode 3.1.2.1 - Configure for use with Pingtel instant xpressa 3.2 - Configure the S8100/G600 3.4 - Configure the VoIP Endpoint(s) – Pingtel instant xpressa
	Cisco SIP IP Phone	3.1 - Configure the Intel PBX IP Media Gateway 3.1.1 - Configure SIP Emulating mode 3.1.2.2 - Configure for use with a Cisco SIP IP Phone 3.2 - Configure the S8100/G600 3.5 - Configure the VoIP Endpoint(s) – Cisco SIP IP Phone

Table 3: Configuration Section Breakdown

3.1. Configure the Intel PBX IP Media Gateway

It is assumed the PBXMG is at factory default settings. For detailed or alternate configuration and parameter information, please refer to the product manual.

1. Verify the PBXMG's physical connections are in place (see **Figure 1**):
   a. LAN port is connected to the network.
   b. Port 1 is connected to port 1 of the TN2224 Digital Line card installed in the S8100/G600 (use an RJ45 cable).
   c. Power cord is connected to an available power source.

2. Configure the Administration PC IP address to **10.12.13.70**, start up a web browser and enter http://10.12.13.74 (default IP address of the PBXMG) to bring up the PBXMG Administration page.

3. Login using an administrative account.

4. Following login, the user is prompted to configure the Operating Mode and PBX Type. Select the Operating Mode and PBX Type desired. For example, for H.323, select **H.323 Emulating** for Operating Mode and **Lucent** for PBX Type (see **Figure 2**). The configuration is similar for SIP.

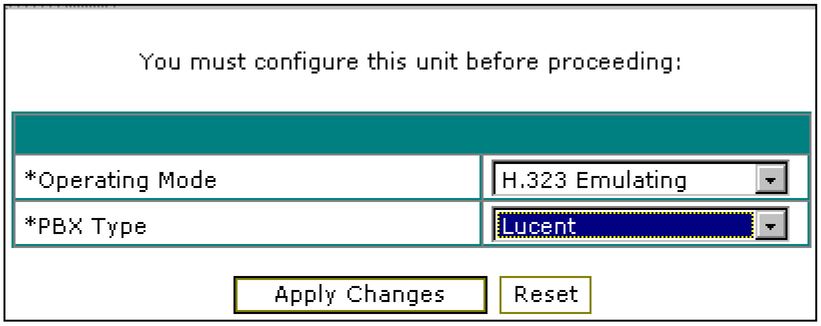

Figure 2: Operating Mode and PBX Type Selection

Click **Apply Changes** to save the settings.

Note 5: Even though the PBX Type lists *Lucent*, it is meant for Avaya equipment.

Note 6: *H.323 Driving* and *SIP Driving* modes relate to the Phone Driving mode supported by model PIMG80PBXDSI and are beyond the scope of this document.

5. Select IP in the left hand menu, set the unit's IP address to **192.16.200.226** and subnet mask to **255.255.255.0** (see **Figure 3**):

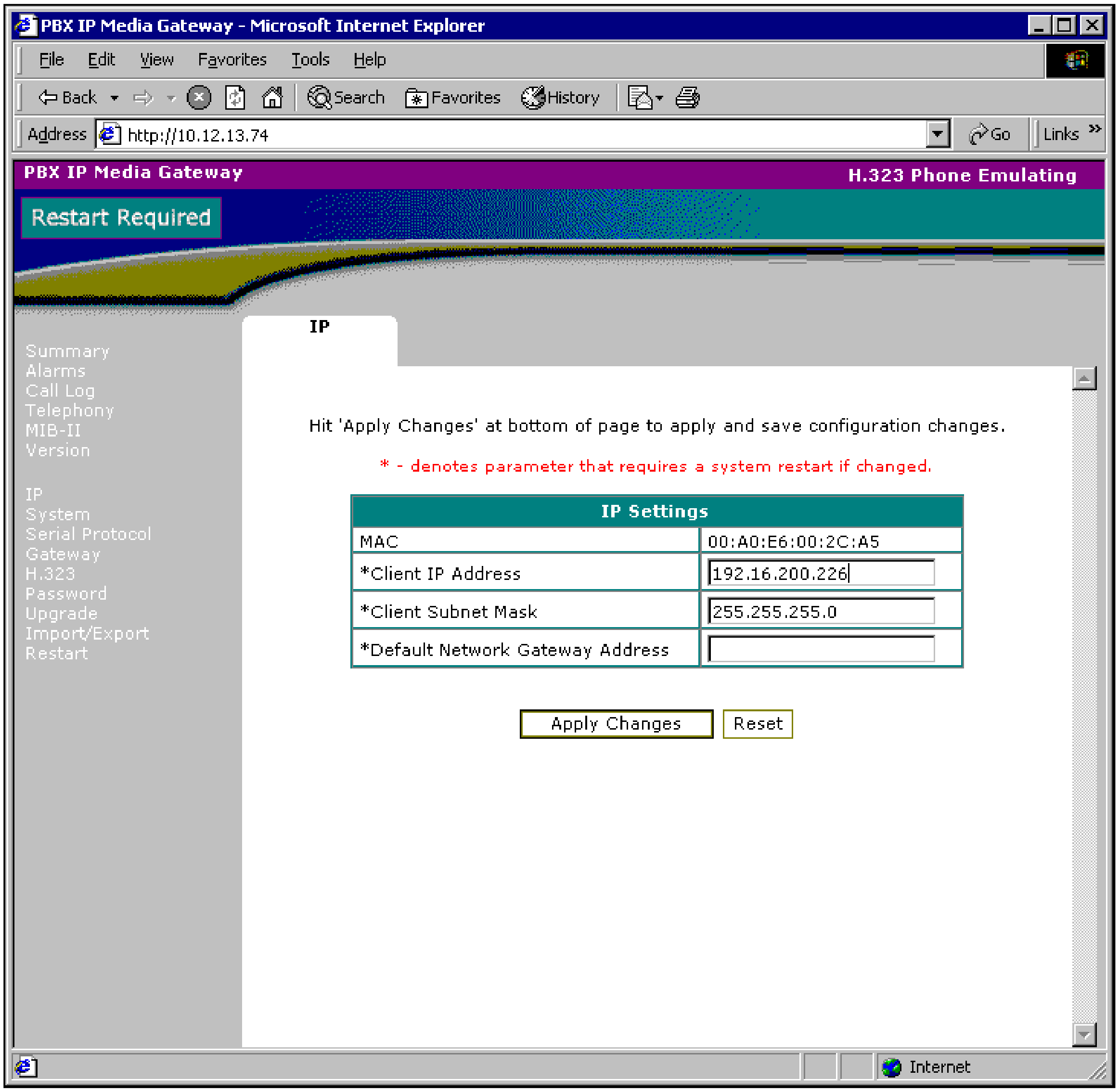

Figure 3: IP Address Configuration

Click **Apply Changes** to save the settings.

6. Restart the unit.

7. Configure the Admin PC IP address to **192.16.200.200**, start up a web browser and enter http://192.16.200.226 to bring up the PBXMG Administration page and login again.

3.1.1. Configure H.323 Emulating mode

1. Select Gateway in the left hand menu, set the following (see **Figure 4**):

   `Call Route Mode` **`Point-To-Point`**
   `Call Connect Mode` **`Instant`**
   `Port 1 Endpoint` **`192.16.200.200`** `(IP of H.323 endpoint)`

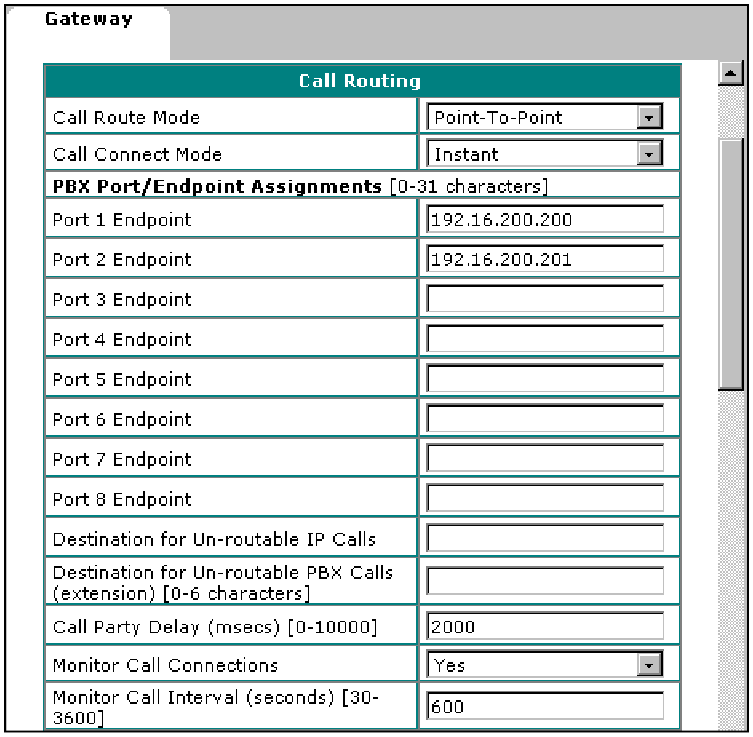

Figure 4: H.323 Gateway Settings

Click **Apply Changes** to save the settings.

Note 7: By default, the IP addresses for Ports 3 through 8 in **Figure 4** would be pre-populated with IP addresses 10.1.1.103 – 10.1.1.108 until locally modified. They were removed in **Figure 4** for clarity.

2. Proceed to *Section 3.2* to configure the S8100/G600 station port and *Section 3.3* to configure an H.323 endpoint such as Microsoft NetMeeting.

3. (*Optional*) To add a second NetMeeting VoIP endpoint with an IP address such as 192.16.200.201:
   - Connect a cable from the S8100/G600 digital line card to port 2 of the PBXMG (Section 3.1, step 1).
   - Enter IP address 192.16.200.201 in the ‘Port 2 Endpoint’ field (Section 3.1.1, step 1).
   - Configure a second station port on the S8100/G600 (Section 3.2).
   - Configure the second NetMeeting endpoint (Section 3.3).

3.1.2. Configure SIP Emulating mode

While the gateway configuration of the PBXMG was identical for the Pingtel instant xpressa and Microsoft NetMeeting, configuration of the Cisco SIP IP Phone required distinct steps to work. The configuration of both endpoints is provided below.

3.1.2.1 Configure for use with Pingtel instant xpressa

1. Select Gateway in the left hand menu, set the following (see **Figure 5**):

```
Call Route Mode       Point-To-Point
Call Connect Mode     Instant
Port 1 Endpoint       192.16.200.200    (IP of SIP endpoint)
```

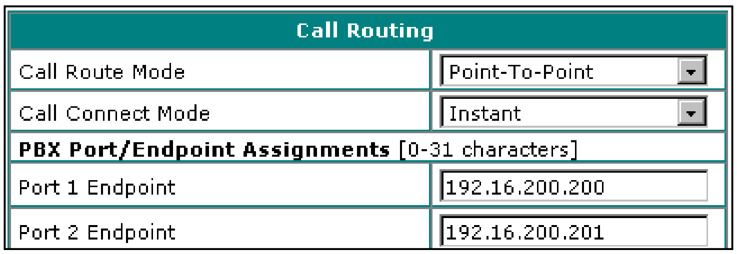

Figure 5: SIP Gateway Settings

   Click **Apply Changes** to save the settings.

2. Proceed to *Section 3.2* to configure the S8100/G600 station port and *Section 3.4* to configure the Pingtel instant xpressa (SIP VoIP endpoint).\

3. (*Optional*) To add a second Instant xpressa VoIP endpoint with IP address 192.16.200.201:
   - Connect a cable from the S8100/G600 digital line card to port 2 of the PBXMG (Section 3.1.1, step 1).
   - Enter IP address 192.16.200.201 in the ‘Port 2 Endpoint’ field (Section 3.1.2.1, step 1).
   - Configure a second station port on the S8100/G600 (Section 3.2).
   - Configure the second instant xpressa endpoint (Section 3.4)

3.1.2.2 Configure for use with a Cisco SIP IP Phone

1. Select Gateway in the left hand menu, set the following (see **Figure 6**):

```
Call Route Mode        Point-To-Point
Call Connect Mode      Instant
Port 1 Endpoint        73314    (ext. assigned to Cisco Phone)
```

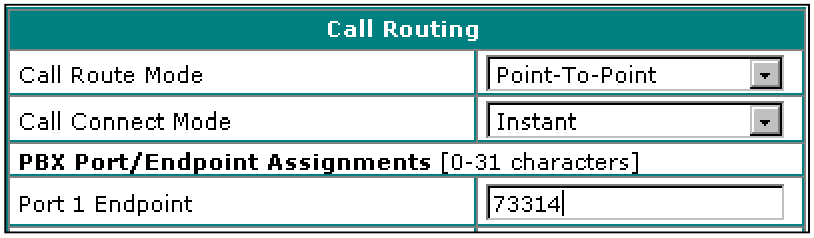

Figure 6: SIP Gateway Settings (Cisco SIP IP Phone)

Click **Apply Changes** to save the settings.

2. Select SIP in the left hand menu, set Proxy Server IP Address to **192.16.200.103** (IP address of the Cisco SIP IP Phone) (see **Figure 7**):

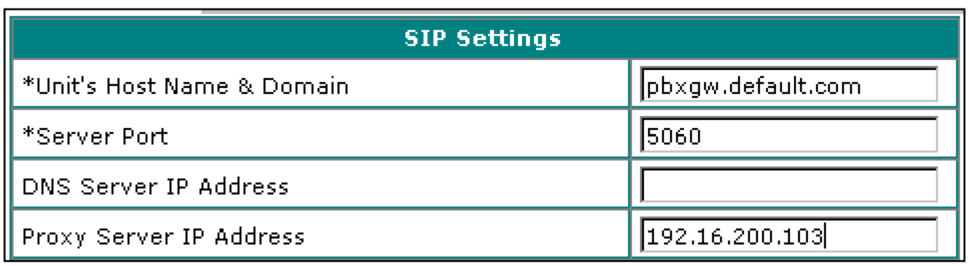

Figure 7: SIP Parameter Settings (Cisco SIP IP Phone)

Click **Apply Changes** to save the settings.

3. Proceed to *Section 3.2* to configure the S8100/G600 station port and *Section 3.5* to configure Cisco SIP IP Phone (SIP VoIP endpoint).

Note 8: This configuration does not permit the use of more than 1 Cisco SIP IP Phone because the endpoint IP address is placed in the Proxy Server IP Address field. According to Intel, this configuration is for verifying that point-to-point calls can work with the Cisco SIP IP Phone. In order to have a configuration with more than one Cisco phone, either a Registration Server or Proxy Server would be required.

3.2. Configure the Avaya S8100/G600

1. Login to the S8100/G600.

2. To add an extension for the port connected to the PBXMG, type **add station 73314** and set the following:

   - Type: **8434D** (see **Figure 8**) – *(Recommended in the Intel User's Guide.)*

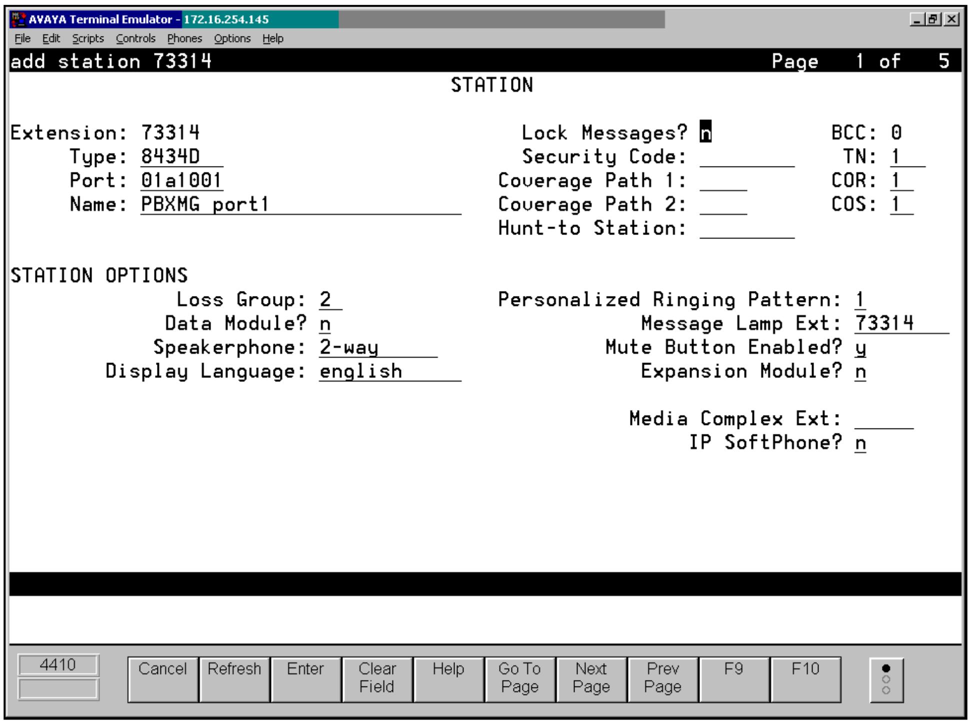

Figure 8: Add Station 73314 – Page 1 of 5

- LWC Reception: **msa-spe** (see **Figure 9**).
  Note 9: *The Intel User's Guide recommends 'y' which may refer to an earlier version of Multivantage software.*

- LWC Activation: **y** (see **Figure 9**)

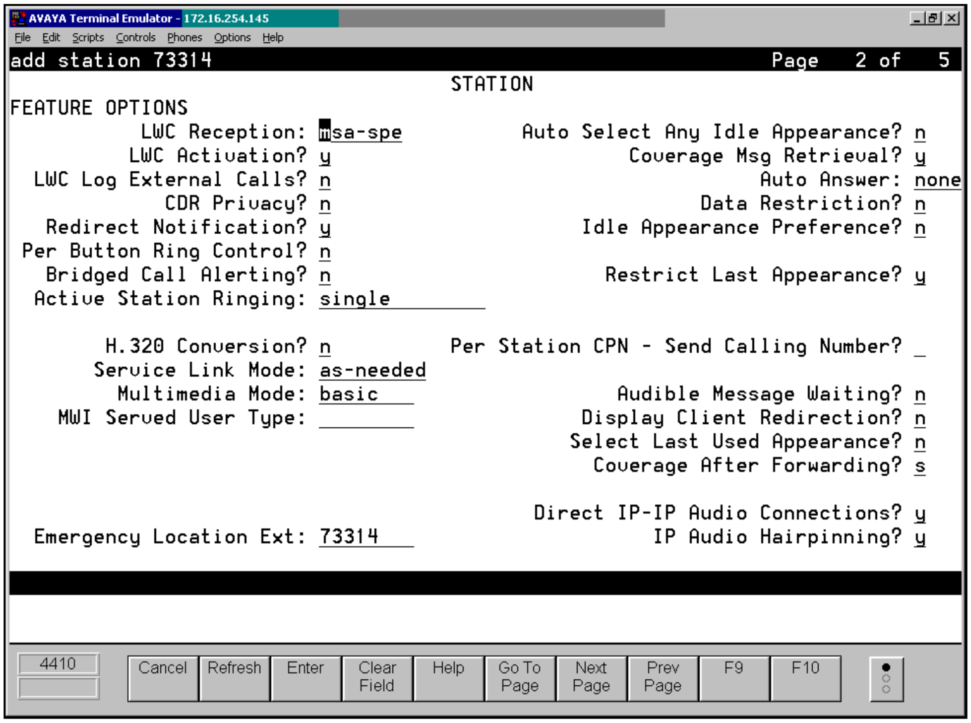

Figure 9: Add Station 73314 – Page 2 of 5

- Assign ‘call-appr’ in the first two button assignments (see **Figure 10**) and clear out all other feature button and soft button settings

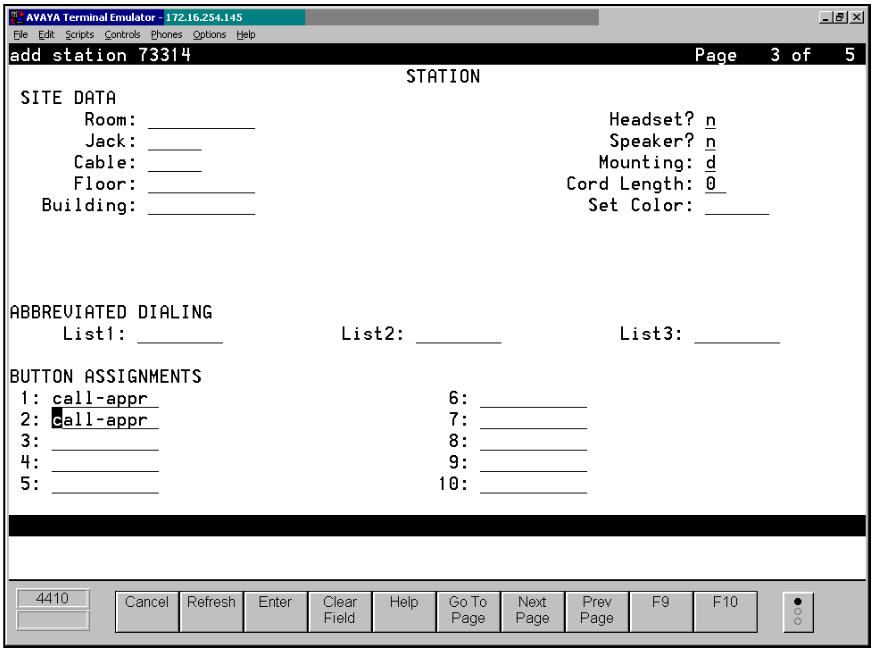

Figure 10: Add Station 73314 – Page 3 of 5

3. To determine the Leave Word Calling Send a Message code and the Leave Word Calling Cancel a Message code, type **display feature-access-codes** (see **Figure 11**). These feature access codes will be used to verify that the Message Waiting Lamp can be turned on or off in the Verification Section (Section 4).

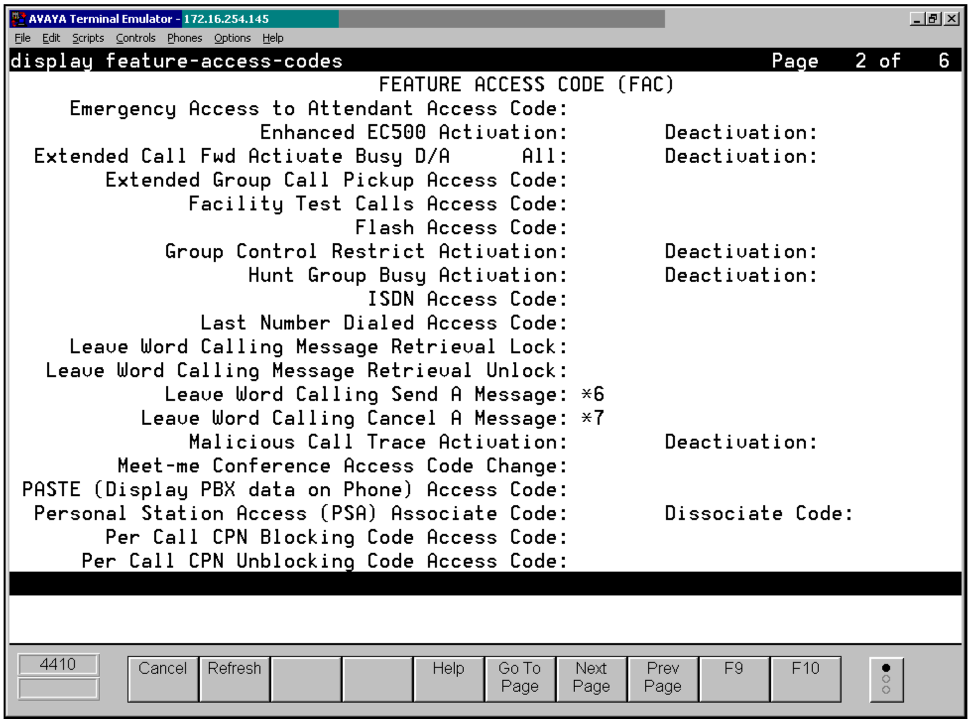

Figure 11: Display Feature Access Code

4. To verify the port is in service, type **status station 73314** (see **Figure 12**) and confirm the Service State is ‘in-service/on-hook’.

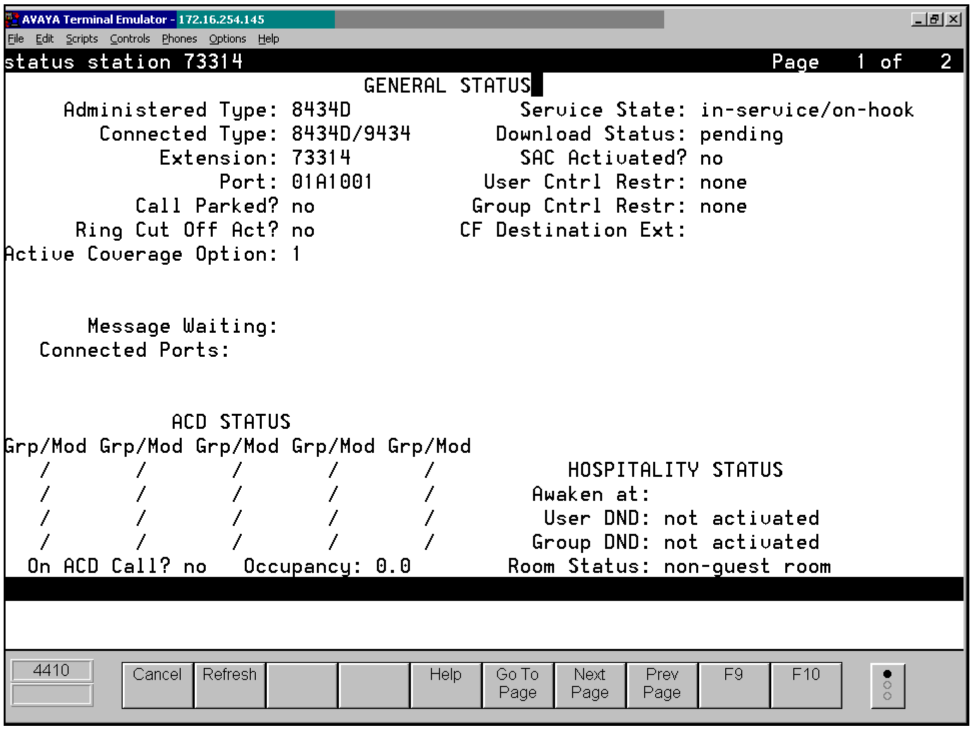

Figure 12: Status Station 73314

5. To verify that the PBXMG port is active, ensure the LED in port 1 of the PBXMG is solid green.

3.3. Configure the VoIP Endpoint(s) - Microsoft NetMeeting

It is assumed Microsoft NetMeeting is at factory default settings prior to configuration.

1. Launch Microsoft NetMeeting on the Admin PC.
2. Select **Tools -> Options** from the pull down menu.
3. In the Options window, click **Advanced Calling** (see **Figure 13**).

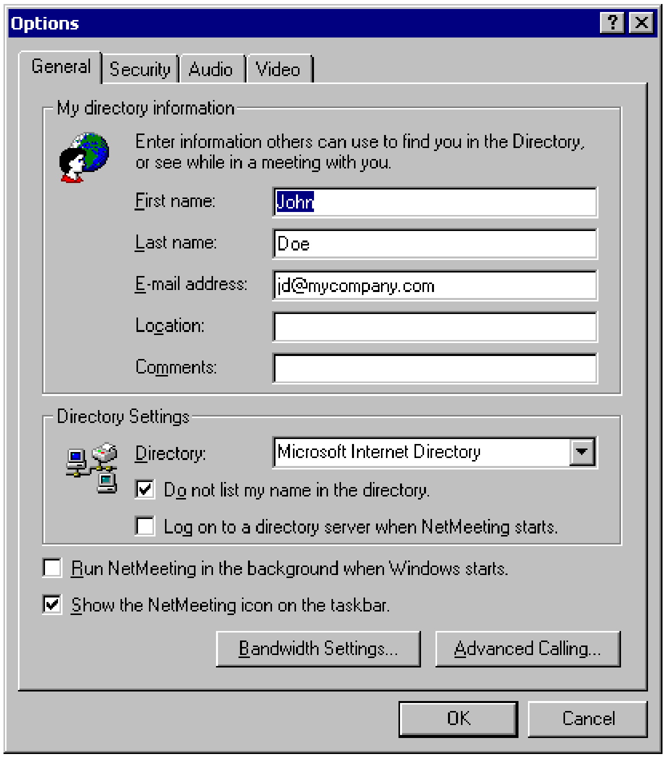

Figure 13: NetMeeting Options

4. In the **Advanced Calling Options** window, check **Use a gateway to call telephones and videoconferencing systems** and set **Gateway** to **192.16.200.226** (see **Figure 14**):

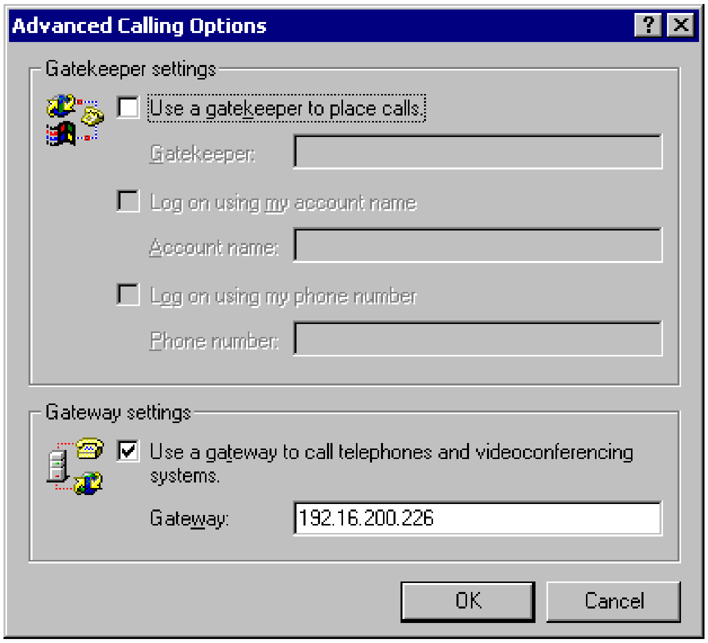

Figure 14: NetMeeting Advanced Calling Options

5. Proceed to *Section 4* for verification steps.

3.4. Configure the VoIP Endpoint(s) - Pingtel instant xpressa

It is assumed Pingtel instant xpressa is at factory default settings prior to configuration.

1. Launch the Pingtel instant xpressa application on the Admin PC.

2. Start up a web browser and enter http://192.16.200.200 to bring up the Pingtel instant xpressa Administration Web Page.

3. Login using an administrative account.

4. Select **Administration -> Phone Configuration** from the top menu bar in the *my xpressa Home* page that appears (see **Figure 15**).

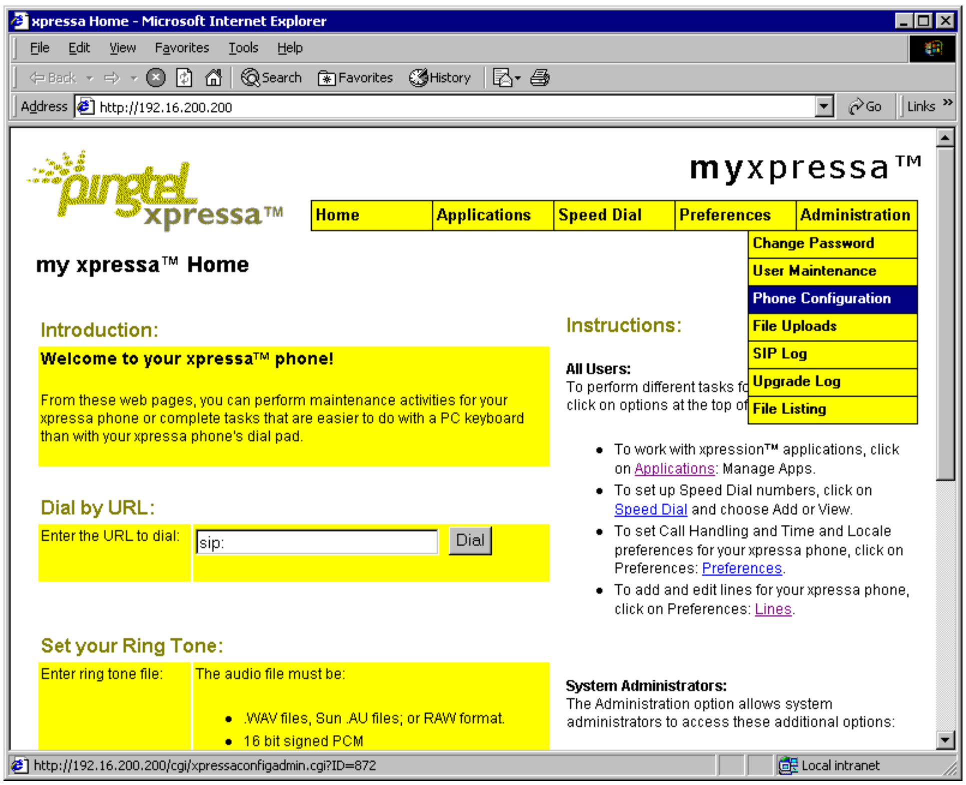

Figure 15: my xpressa Home page

5. In the Phone Configuration page, scroll down to the *Call Addressing* section and set the following in the digitmap box (see **Figure 16**):

```
xxxxx : sip:{digits}@192.16.200.226
```

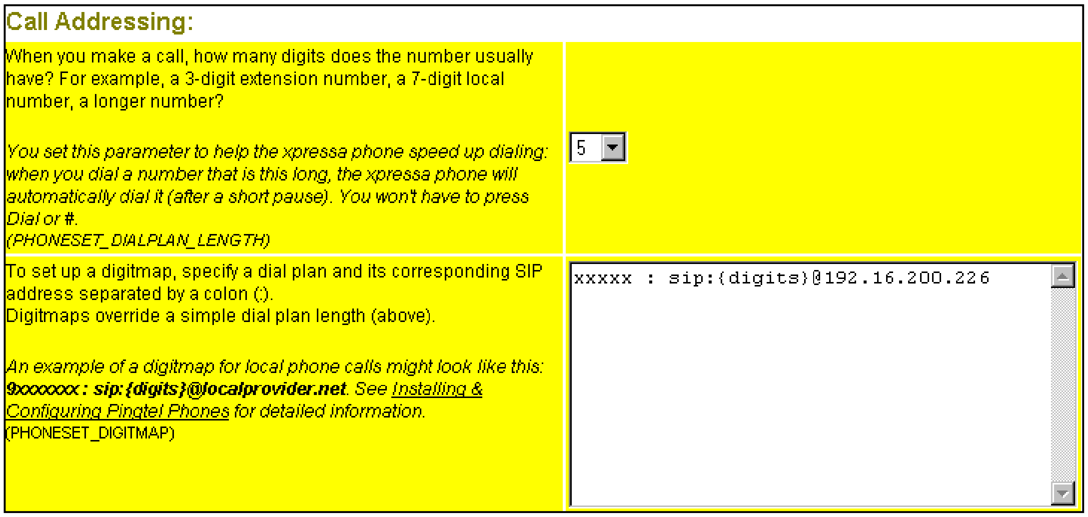

Figure 16: my xpressa: Call Addressing

Click **Save** then click **Restart**.

When the web browser displays “Restarting phone…” go to the instant expressa application window and click **Ok** to restart now (see **Figure 17**).

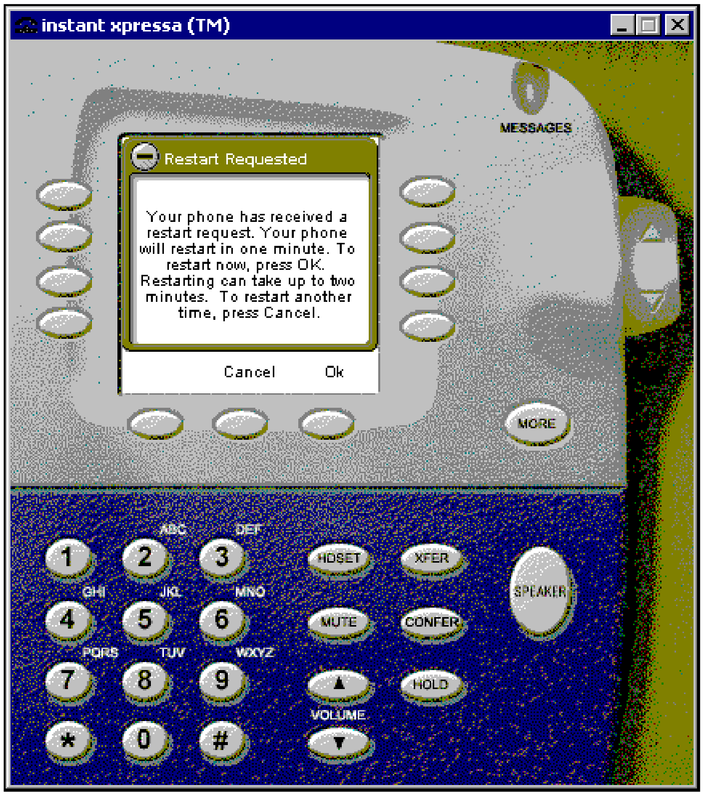

Figure 17: instant xpressa Application Restart

6. Following application restart, return to the main browser page, select **Preferences-> Lines** from the top menu bar (see **Figure 18**).

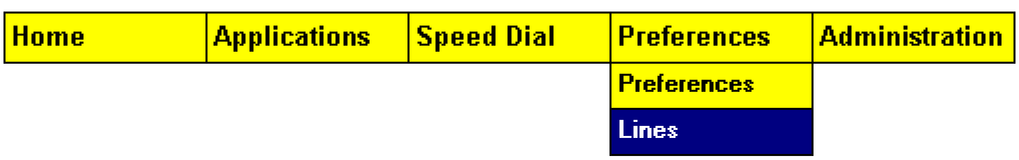

Figure 18: Preferences -> Lines

7. In the Multiple Lines window, edit the Device Line entry and set the SIP URL to **sip: 73314@192.16.200.200**.

8. Proceed to *Section 4* for verification steps.

3.5. Configure the VoIP Endpoint(s) - Cisco SIP IP Phone

1. Press the **settings** button on Cisco SIP IP Phone keypad.

2. Scroll down to *SIP Configuration* and press the **Select** soft button.

3. If the icon next to the SIP Configuration title appears to be a closed lock, press ****#** to unlock the phone. If the icon appears as an open lock, proceed to the next step.

4. Choose *Line 1 Settings* and press **Select**.

5. In Line 1 Configuration screen, choose the Name field, press **Edit**, enter **73314** as the New Name, and then press **Accept**.

6. In the Line 1 Configuration screen, scroll down to the Proxy Address field, press **Edit**, enter **192.16.200.226** (IP address of the PBXMG) as the New Proxy Address, and then press **Accept**.

7. In the Line 1 Configuration screen, press **Back** to return to the SIP Configuration screen.

8. In the SIP Configuration screen, scroll down to the Register with Proxy field. If the word YES appears next to the Register with Proxy Field, press **No**.

9. While still on the SIP Configuration screen, scroll further down to the Outbound Proxy field, press **Edit**, enter **192.16.200.226** (IP address of the PBXMG) as the New Outbound Proxy, then press **Accept**.

10. In the SIP Configuration screen, press **Save**.

11. Proceed to *Section 4* for verification steps.

4. Verification Steps

The following steps can be used to verify the configuration depicted in **Figure 1**:

- Verify a green status light appears when the PBXMG telephony port is connected to the S8100/G600 port.
- Verify station 73314 is online by executing **status station 73314** on the S8100/G600.
- Verify ping to the PBXMG and the VoIP endpoint for each given scenario is ok.

Scenario A verification (see Figure 1):

DCP <-> S8100/G600 <-> PBXMG <-> MS NetMeeting

- DCP Phone (x73310) call to NetMeeting PC (x73314)
- NetMeeting PC (x73314) call to DCP Phone (x73310)
- DCP Phone (x73310) call to NetMeeting PC (x73314)
- NetMeeting PC (x73314) call to DCP Phone (x73310)

Scenario B verification (see Figure 1):

DCP <-> S8100/G600 <-> PBXMG <-> Cisco SIP

- DCP Phone (x73310) call to Cisco SIP IP phone (x73314)
- Cisco SIP IP Phone (x73314) call to DCP Phone (x73310)
- DCP Phone (x73310) call to Cisco SIP IP phone (x73314)
- Cisco SIP IP Phone (x73314) call to DCP Phone (x73310)
- Dial FACs (*6/*7) from the DCP phone (x73310) to turn on/off the Message Waiting lamp on Cisco SIP IP Phone (x73314)

Scenario C verification (see Figure 1):

DCP <-> S8100/G600 <-> PBXMG <-> Pingtel Instant xpressa

- DCP phone (x73310) call to Instant xpressa PC (x73314)
- Instant xpressa PC (x73314) call to DCP phone (x73310)
- DCP phone (x73310) call to Instant xpressa PC (x73314)
- Instant xpressa PC (x73314) call to DCP phone (x73310)
- Dial FACs (*6/*7) from the DCP phone (x73310) to turn on/off the Message Waiting lamp on Instant xpressa application

5. Conclusion

The Intel PBX IP Media Gateway can be used in limited point-to-point calling scenarios with the Avaya S8100 Media Server / G600 Media Gateway and either H.323 or SIP endpoints. The PBX IP Media Gateway does not provide emulation support for both types of endpoints simultaneously. These Application Notes illustrate how to set up basic point-to-point calling between the Intel PBX IP Media Gateway and the Avaya S8100 Media Server / G600 Media Gateway. Use of the Cisco SIP IP phone in this configuration is too limited for uses beyond verifying that point-to-point calls are possible because the configuration requires that the Intel PBX IP Media Gateway's Proxy Server IP Address field be configured with the IP address of the Cisco SIP IP phone. According the Intel, this would not be required if a SIP Registration Server or Proxy Server were used.

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