This Quick Setup guide is intended for technicians installing Avaya 4602 SIP telephones.

Before performing the procedures in this guide, make sure the following conditions exist:

- An Avaya Communication Manager (CM) Release 2.1.1 with field load 415 is installed and running.
- An Avaya Converged Communications Server (CCS) Release 2.1 is installed and running.
- DNS is administered properly (if being used by the customer).
- DHCP scope 172 has been created on the DHCP server. Under DHCP Scope Options, set Option 172 to **ConfigHttpSrvr=a.b.c.d**, where **a.b.c.d** is the address of the HTTP server.
- You are familiar with administering Avaya Communication Manager and Avaya Converged Communications Server.

When setting up SIP trunks, you will need to know the following information:

- number of off-PBX stations
- maximum number of SIP trunks
- node name of the Avaya Converged Communications Server
- IP address of the Avaya Converged Communications Server
- TAC for the SIP trunk group
- number of members for the SIP trunk group

Release 1.0 to Release 1.1 Migration Notes

In Release 1.0 of the 4602 SIP Telephone, the extension name of the 4602 SIP telephone had to be different from the extension name used in Avaya Communication Manager. In Release 1.1 (which is supported by Avaya Communication Manager Release 2.1.1 and Avaya Converged Communications Server Release 2.1), the extension name of the 4602 SIP telephone must be identical to the extension name used in Avaya Communication Manager.

- Changes on Avaya Communication Manager
  - Only OPS administered SIP endpoints are supported in Avaya Communication Manager Release 2.1.1. “Pure SIP” or standalone configurations are not supported.
  - Hotline is not supported.
  - Assign an AWOH station number for the SIP/OPS endpoints (if not done already). (Only AWOH is supported in Avaya Communications Manager Release 2.1.1.)
  - Update the off-pbx-telephone station mapping so that the station names of the AWOH and the 4602 SIP telephone match.
- Changes on Avaya Converged Communications Server
  - Update existing Avaya Converged Communications Server user records to include a Media Server Extension. The Media Server Extension should match the AWOH station assigned in Avaya Communication Manager.
  - If the user handle is an extension, it must match the AWOH station assigned.
- Changes on the 4602 SIP Telephone
  - Change the extension name so that it matches the AWOH station name on Avaya Communication Manager.
  - Remove the Hotline option and Extension in the Call Handling page.
  - Enter the registration domain name in the configuration files so that the 4602 SIP Telephone can register with Avaya Converged Communications Server Release 2.1.

1

Configure the Avaya Communication Manager.

1. Use the **change system-parameters customer options** command to set the following parameters:
   - the maximum number of off-PBX telephones - OPS (page 1 of form)
   - the maximum number of SIP trunks (page 2 of form)
   - Enhanced EC500 to “y” (page 4 of form)
   - IP trunks to “y” (page 4 of form)

   Important:

   These parameters are determined by the license file. Only an Avaya-authorized representative can modify these parameters.
2. Use the **change node-names ip** command to set the following parameters:
   - host name of the Avaya Converged Communications Server
   - IP address of the Avaya Converged Communications Server
3. Use the **change system-parameters features** command to set Trunk-to-Trunk transfer to “all.”
4. Use the **change ip-network-region** command to set the home domain (domain.com) of the Avaya Converged Communications Server on all applicable regions.

2

Set up a SIP trunk between the Avaya Communication Manager and the Avaya Converged Communications Server.

1. Use the **add signaling-group #** command to add a SIP signaling group. Set the following parameters:
   - Group Type to “sip”
   - Transport Method to “tls”. (This is the default setting.)
   - Depending on the type of system, perform one of the following steps:
     - For an S8500 system or an S8700 system, set Near-end Node Name to a designated CLAN.
     - For an S8300 system, set Near-end Node Name to “procr.”
   - Far-end Node Name. Enter the node name of the Avaya Converged Communications Server you entered in Step 2 of Procedure 1.
   - Near-end Listen Port to “5061”. (This is the default setting.)
   - Far-end Listen Port to “5061”
   - Far-end Domain Name. Enter either the local Avaya Converged Communications Server domain name or the far-end SIP server domain name to be reached.
   - DTMF over IP to “rtp-payload”. (This is the default setting.)
2. Use the **add trunk-group #** command to add a SIP trunk group. Set the following parameters:
   - Group Type to “sip”
   - Group Name
   - TAC. (You must enter a TAC even though the TAC is not used for accessing a SIP trunk.)
   - Service Type to “tie”
   - Signaling Group to the number of the signaling group you created in Step 1 of this procedure.
   - Number of Members (up to 255). (This is the number of concurrent users who will be using the trunk.)
   - Administer fields on page 2 as appropriate for the customer.

Step	Description
2 **cont.**	3. Use the **change route-pattern #** command to route outgoing calls over the SIP trunks by using AAR/ARS. Set the following parameters: - Trunk Group Number to the number of the trunk group you added in Step 2. - FRL to “0” or appropriate FRL number - A route-pattern name that is easy to identify. **Note:** Even though AAR/ARS (and therefore route patterns) is optional in Avaya Communication Manager R2.1.1, you need a route pattern administered for next step. 4. Use the **change locations** command to specify the route pattern. Set the following parameters: - Proxy Sel Rte Pat. to the route pattern you added in Step 3. - Make the same route pattern entry for each location administered on this form. 5. If using AAR, use the **change aar analysis #** command to specify how Avaya Communication Manager will analyze a digit string to determine the pattern for the call. Set the following parameters: - Minimum/maximum digits to be used in the trunk call - Route pattern #
3	Use the **add station #** command to create an AWOH station for the 4602 SIP phone on the Avaya Communication Manager. (Only AWOH stations are supported in the Avaya Communication Manager Release 2.1 offer.) Use the default settings for all fields except the Port field. Enter **X** in the Port field.
4	Configure an OPTIM extension on the Avaya Communication Manager. 1. Use the **add off-pbx-telephone station-mapping** command to configure an OPTIM extension. Set the following parameters: - Station Extension to the extension of the 4602 SIP phone. (You added this extension in Procedure 3.) - Application to “OPS” - Dial Prefix to the AAR/ARS code used for the SIP trunk if AAR/ARS is being used. Otherwise, this field is blank. - Phone Number to the extension administered on the Avaya Converged Communications Server - Set Trunk Selection to either “aar,” “ars,” or the trunk number of the SIP trunk. (This is the trunk group number you specified in Step 2 of Procedure 2.) It is recommended that you use the trunk number. - Configuration Set. Enter **1** if you want to use the default settings. Be sure to review the fields under the configuration set to customize for each customer. For customers that do not use the North American dial plan, Fast Connect on Origination should be set to “Yes” to receive Avaya Communication Manager call progress tones. 2. If extended features will be used, use the **change off-pbx-telephone feature-name-extensions** command to assign extensions to available features (such as conferencing, call forwarding, and send all calls).

5

Configure the Avaya Converged Communications Server.

1. Create a Media Server. (The Media Server provides a connection between the Avaya Converged Communications Server and the Avaya Communication Manager using a designated CLAN/procr.)
2. Create Address Maps for the Media Server you created in Step 1.

   These address maps will be used to represent Avaya Communication Manager dial plans.

   For example, the map ^sip:123[0-9]{2}@avaya.com will map extensions 12300 to 12399 (inclusive) to the media server contact. The map ^sip:91732[0-9]{7}@avaya.com will allow users to dial a telephone number in the 732 area code.

   Address maps will be needed for all call types (911|local extension dialing|local outside|long distance|international).
3. Add a user for the 4602 SIP phone. Set the following parameters:
   - User Handle. Enter a name or number. (While you can enter a number, it is recommended that you enter a name. The media server extension will resolve which devices will ring for the selected user.)
   - Password. The password is alphanumeric.
   - Host to which the user will register
   - First Name of user
   - Last Name of user
   - Add Media Server Extension. The Media Server Extension should match the AWOH extension created on Avaya Communication Manager.

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Modify and copy the configuration files to the appropriate servers. You can download the files in this step from www.avaya.com/support.

1. Save your changes, and then copy the file "46xxupgrade.scr" to the TFTP server.
2. Copy the file "323tosip0xxx.bin" to the TFTP server. (This file instructs the 4602 telephone to change from H.323 to SIP.)
3. Copy the file "sip_4602bt0xxx.ebin" to the root directory of the HTTP server.
4. Copy the file "sip_4602ap0xxx.ebin" to the root directory of the HTTP server.
5. Copy the following files to the root directory of the HTTP server:
   - sip_4602D01A.txt (This file will be used by 4602 telephones.)
   - sip_4602D02A.txt (This file will be used by 4602SW telephones.)
6. Open the file "sip_4602D01A.txt" and perform the following steps on the specified lines:
   - #ConfigHttpSrvr

     Delete "#" and enter the IP address of the HTTP server.
   - #ProxyServers

     Delete "#" and enter the proxy server.
   - #RegistrarServers

     Delete "#" and enter the registrar servers.
   - #SiteOption

     Delete "#" and enter **172**. (This is the DHCP scope that was created previously on the DHCP server for the SIP telephones.) For information on how to set up a DHCP server, see the section "Setting Parameters" in the *Avaya 4602 SIP Telephone Release 1.1 Administrator's Guide*, 16-300037, Issue 2, September 2004.
   - #DialPlan

     Delete "#" and enter the dial plan. For information on how to configure a dial plan, see the section "Configuring a Dial Plan" in the *Avaya 4602 SIP Telephone Release 1.1 Administrator's Guide*, 16-300037, Issue 2, September 2004

6

cont.

7. When finished, save your changes and then exit the file.
8. Open the file “sip_4602D02A.txt” and perform the following steps on the specified lines:
   - #ConfigHttpSrvr
     Delete “#” and enter the IP address of the HTTP server.
   - #ProxyServers
     Delete “#” and enter the proxy server.
   - #RegistrarServers
     Delete “#” and enter the registrar servers.
   - #SiteOption
     Delete “#” and enter **172**. (This is the DHCP scope that was created previously on the DHCP server for the SIP telephones.)
   - #HotLine
     Delete “#” and enter **0**.
9. When finished, save your changes and then exit the file.

7

Convert the 4602 IP phone from H.323 to SIP, and then administer the telephone. (The conversion from H.323 to SIP will take approximately five minutes for each telephone.)

1. Connect the 4602 IP phone to the system.
2. At the 4602 IP phone, press ***** when `Press * to program` appears on the display of the telephone.
3. Press **#** to go through all of the settings and access a blank screen. Do not change any of these settings.
4. At the blank screen, press **MUTE 7 4 4 #**.

   `Sig=default` appears on the display of the telephone.
5. Press ***** until `Sig=SIP` appears on the displays of the telephone.
6. Save the changes.
7. Press **MUTE 7 3 73 8 #**. to reset the telephone. Do not reset the values.

   The telephone will convert from H.323 to SIP. After approximately five minutes, the following message will appear on the display of the telephone:

   No Service
8. Press **MUTE 4 6 3 6 #**.

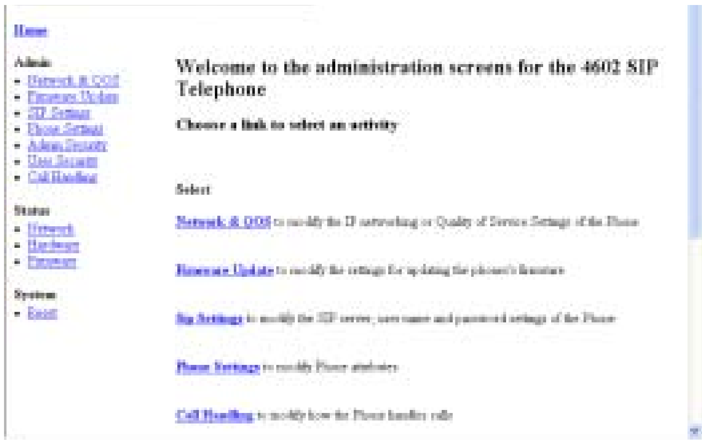
   The IP address of the telephone is displayed. (You may want to write down this IP address.)
9. Start your web browser, enter the IP address of the 4602 SIP phone (from Step 2), and press ENTER.

   The Login dialog box appears.

7 cont.

10. In the User Name box, type **admin**.
11. In the Password box, type **barney**.
12. Click the **OK** button.

    The 4602 home page appears.



13. Click **SIP Settings**.

    The SIP Settings page appears.

7 cont.

14. In the Name box, enter the AWOH extension assigned in Avaya Communication Manager.
15. In the Password box, enter the password for this extension.
16. In the Messaging Extension box (in the Server Setup area at the bottom of the page), enter the extension on Avaya Communication Manager to which this phone will subscribe for Messaging Waiting indication.
17. Click the **Save** button.
18. Click **Phone Settings**.

    The Phone Settings page appears.

19. In the Display Name box, enter the calling name for this phone (that is, the name that will appear on the display of the called phone when this phone makes a call).
20. Make any other changes.
21. When finished, click the **Save** button.
22. Click **Reset** for the changes to take effect for the 4602 SIP phone.

    After the phone is converted successfully, `SIP extension` appears on the display of the telephone.

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Repeat Procedure 7 for every 4602 SIP telephone.