Avaya Solution & Interoperability Test Lab

Application Notes for Bose ControlSpace EX-Series Conferencing Processor with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Bose ControlSpace EX-Series Conferencing Processor with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Bose ControlSpace EX-Series Conferencing Processor is an audio-conferencing solution that registers with Avaya Aura® Session Manager via SIP. For this compliance test, the Bose ControlSpace EX-1280C Conferencing Processor was used. In addition, the dialer in the Bose ControlSpace Designer and the Bose ControlSpace Remote were used for managing calls.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to integrate the Bose ControlSpace EX-Series Conferencing Processor with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Bose ControlSpace EX-Series Conferencing Processor is an audio-conferencing solution that registers with Avaya Aura® Session Manager via SIP. For this compliance test, the Bose ControlSpace EX-1280C Conferencing Processor was used. In addition, the dialer in the Bose ControlSpace Designer and the Bose ControlSpace Remote were used for managing calls.

The ControlSpace EX-Series also includes the EX-440C, as detailed in **Attachment 1**. Since the EX-1280C and EX-440C share the same firmware version and only differ in the number of SIP lines supported, these Application Notes also apply to the EX-440C.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the ControlSpace EX-1280C, Avaya SIP and H.323 Deskphones, and the PSTN. The dialer in the Bose ControlSpace Designer and the Bose ControlSpace Remote were used for managing calls. Telephony features, such as call forward, call coverage, and call pickup were also verified using Communication Manager Feature Access Codes (FACs).

The serviceability testing focused on verifying that the ControlSpace EX-1280C came back into service after re-connecting the Ethernet cable or rebooting the server.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and ControlSpace EX-1280C Conferencing Processor did not include use of any specific encryption features as requested by Bose.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of ControlSpace EX-1280C with Session Manager.
- Calls between ControlSpace EX-1280C and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the ControlSpace EX-1280C and the PSTN.
- G.711 and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, call transfer, and 3-party conference initiated from an Avaya SIP/H.323 Deskphone.
- Extended telephony features using Communication Manager FACs for Last Number Dialed, Call Forward, and Call Pickup.
- Managing calls using the dialer in the ControlSpace Designer and ControlSpace Remote.
- Multiple calls to ControlSpace EX-1280C using separate SIP lines.
- Proper system recovery after a restart of the ControlSpace EX-12980C server and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation(s):

- ControlSpace EX-1280C doesn't support hold, call transfers, or conferencing using the dialer of the ControlSpace Designer or ControlSpace Remote.
- Incoming calls to the ControlSpace EX-1280C displays *UNKNOWN* on the ControlSpace Remote dialer. However, the caller's name and number are displayed on the dialer of the ControlSpace Designer.
- Although ControlSpace EX-1280C only supports one call per SIP line, it is possible for a second call to be attempted using either dialer, which prevents in the first (original) call from being resumed.
- An established call between ControlSpace EX-1280C and Avaya IP phone cannot be dropped from the dialer of the ControlSpace Remote, if the Avaya IP phone places the call on hold. When the Avaya IP phone places the call on hold, the call button on the dialer changes from red, indicating active call can be dropped, to green, indicating a new call can be made. Calls can only be dropped when the call button is red.
- Calls on SIP line 2 cannot be auto-answered even if the feature is enabled. The call continues to ring.
- When an outgoing call fails for whatever reason (e.g., invalid number, blocked call, busy), the dialer in the ControlSpace Remote doesn't provide a failure reason. However, the call button goes from red to green indicating that the call didn't complete. The dialer in the ControlSpace Designer does provide a failure reason in the display section.

2.3. Support

For technical support and information on ControlSpace EX-Series Conferencing Processors, contact Bose Product Support at:

- Phone: 1-800-994-BOSE
- Website: https://pro.bose.com/en_us/contact_pro/contact_us.html

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging serving as the voicemail system.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J129 SIP Deskphones.
- Bose ControlSpace EX-1280C with ControlSpace Designer and ControlSpace Remote.

Bose ControlSpace EX-1280C registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

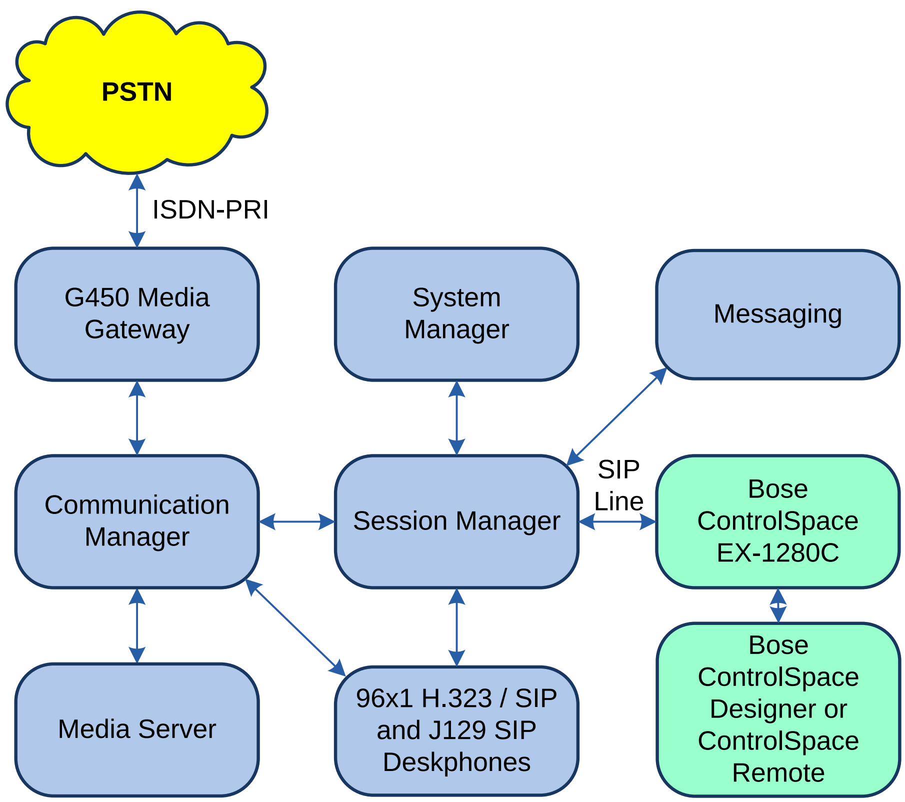


Figure 1: Avaya SIP Network with Bose ControlSpace EX-1280C Conferencing Processor

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.0.1.0-SP1 (R018x.01.0.890.0 with Patch 25393)
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.1.121
Avaya Aura® System Manager	8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.079814
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya 96x1 Series IP Deskphones	6.8003 (H.323) 7.1.5.0.11 (SIP)
Avaya J129 SIP Deskphones	4.0.2.0.8
Bose ControlSpace EX-1280C Conferencing Processor with: ▪ VoIP Application ▪ VoIP Library	FW v2.120 build 1 v1.18.0 v4.4.4
Bose ControlSpace Designer	v5.6.2.30797
Bose ControlSpace Remote	2.7.0.30748

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for the Bose ControlSpace EX-1280C is configured through System Manager in **Section 6.2**.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                      Page   1 of  12
                                OPTIONAL FEATURES

     G3 Version: V18                             Software Package: Enterprise
       Location: 2                                System ID (SID): 1
       Platform: 28                               Module ID (MID): 1

                                                           USED
                            Platform Maximum Ports:  48000     82
                                  Maximum Stations:  36000     21
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:  41000      0
                Maximum Off-PBX Telephones -   OPS:  41000     10
                Maximum Off-PBX Telephones - PBFMC:  41000      0
                Maximum Off-PBX Telephones - PVFMC:  41000      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                     Maximum Survivable Processors:  313        0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Aura® Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avaya.com
    Name:                        Stub Network Region: n
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: 50999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to ControlSpace EX-1280C. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. ControlSpace EX-1280C was tested using G.711 and G.722 codecs. Note that media encryption was configured for Avaya IP Deskphones. ControlSpace EX-1280C didn't support SRTP; hence, the entry with *none* under **Media Encryption**.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

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

     Media Encryption                  Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: none
 3:
 4:
 5:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- Set the **Transport Method** field to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                        Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Bose ControlSpace EX-1280C Conferencing Processor and Avaya SIP Deskphones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                            Page   1 of  22
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm                COR: 1        TN: 1        TAC: 1010
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                             Member Assignment Method: auto
                                                      Signaling Group: 10
                                                    Number of Members: 10
```

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern 10 as shown below.

```
change aar analysis 78                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed            Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    78                    5    5     10        lev0          n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                          Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 10   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 Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

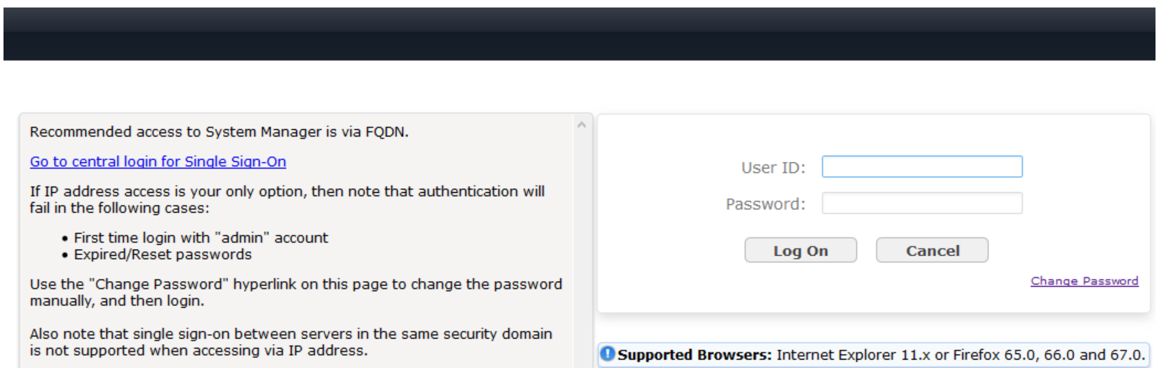
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Bose ControlSpace EX-1280C Conferencing Processor
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for the Bose ControlSpace EX-1280C Conferencing Processor.

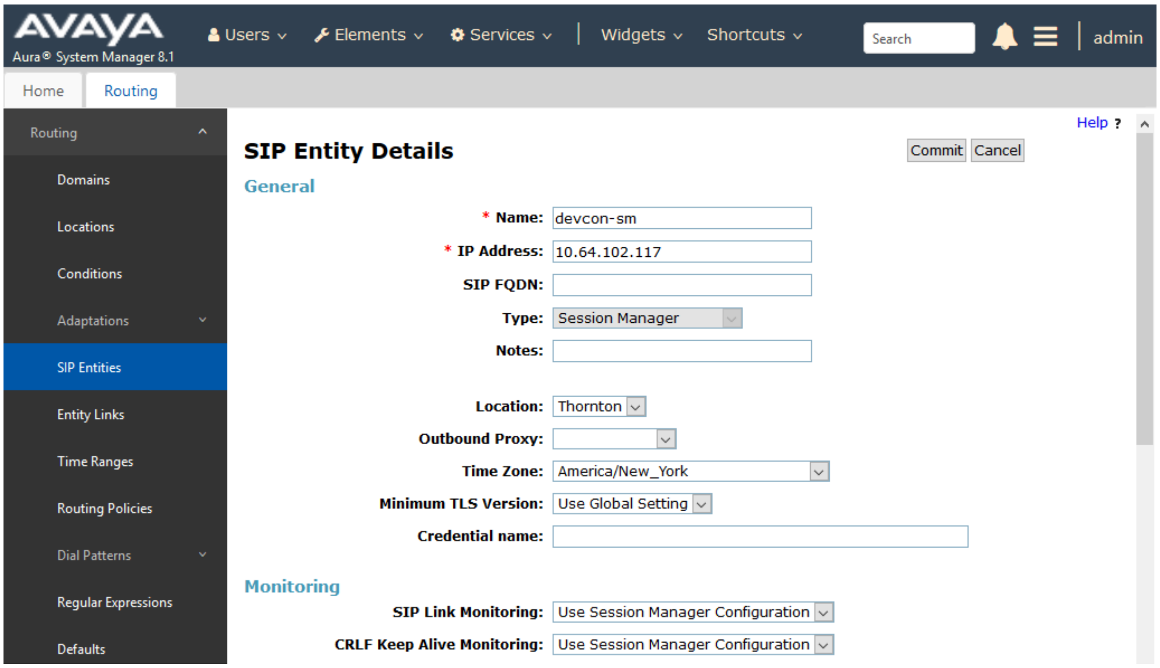
6.1. Launch System Manager

Access the System Manager Web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.

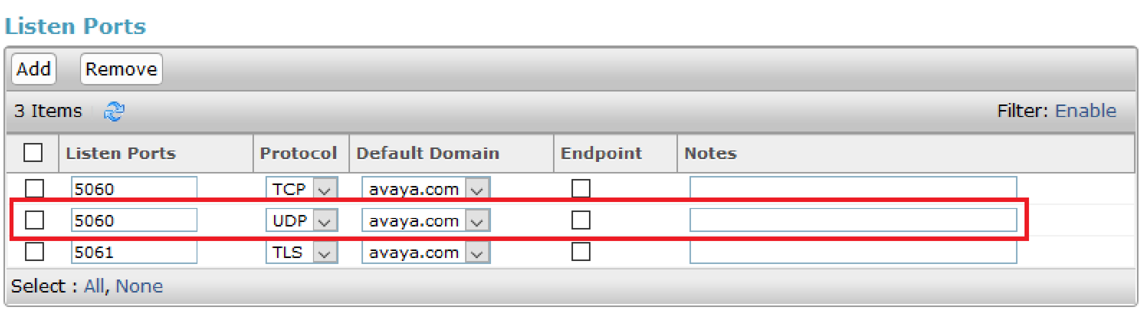


6.2. Set Network Transport Protocol for Bose ControlSpace EX-1280C Conferencing Processor

From the System Manager **Home** screen, select **Elements → Routing → SIP Entities** and edit the SIP Entity for Session Manager shown below.



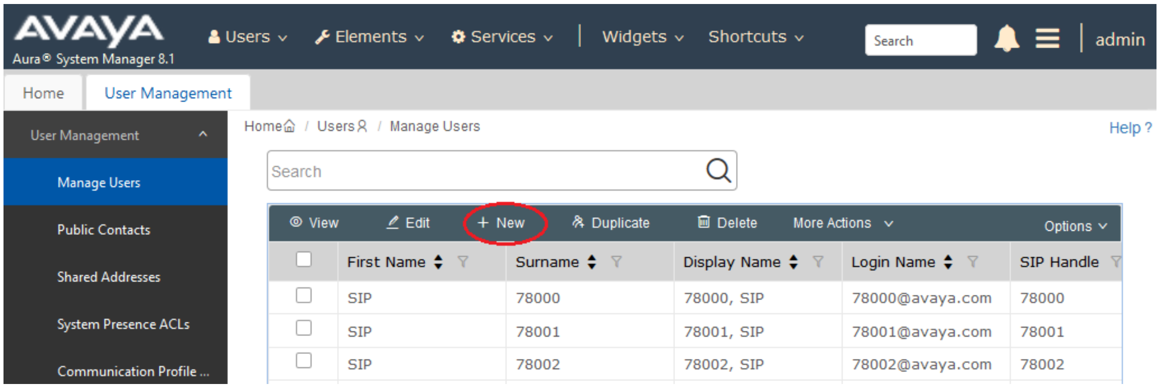
Scroll down to the **Listen Ports** section and verify that the transport network protocol used by ControlSpace EX-1280C is specified in the list below. For the compliance test, the solution used UDP network transport.



6.3. Administer SIP User

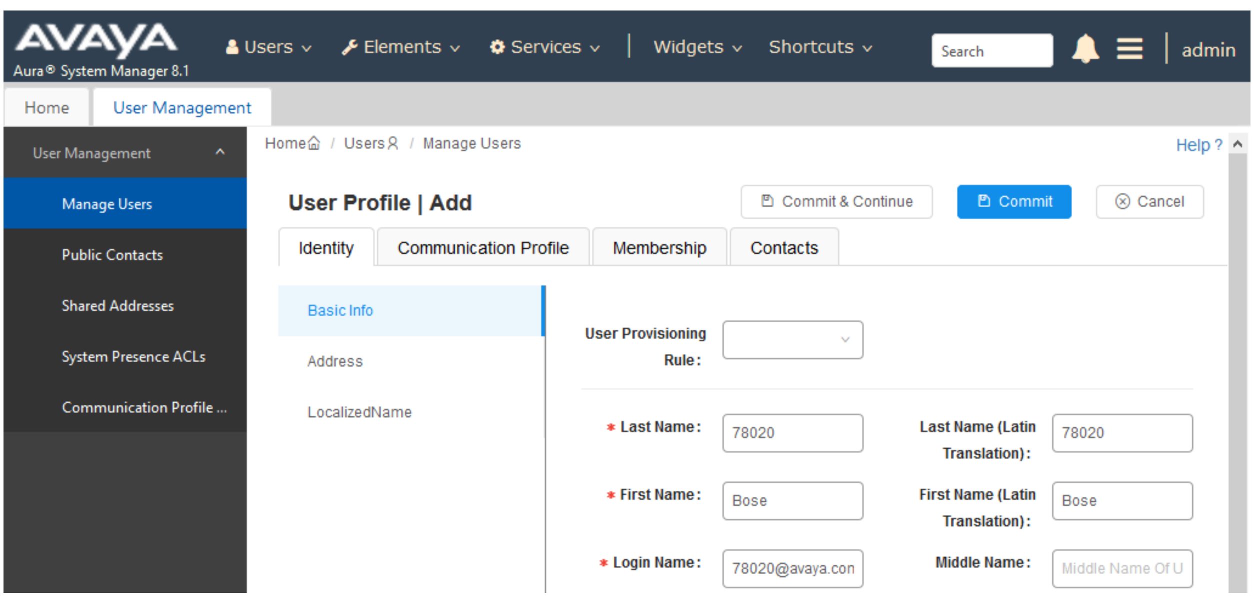
In the subsequent screen (not shown), select **Users → User Management → Manage Users** to display the **User Management** screen below. Click **New** to add a user. This section will show the configuration of the first SIP line with extension 78020.

Note: Repeat the steps in this section for the second SIP line, if desired. For the compliance test, a second SIP line was configured with extension 78021.



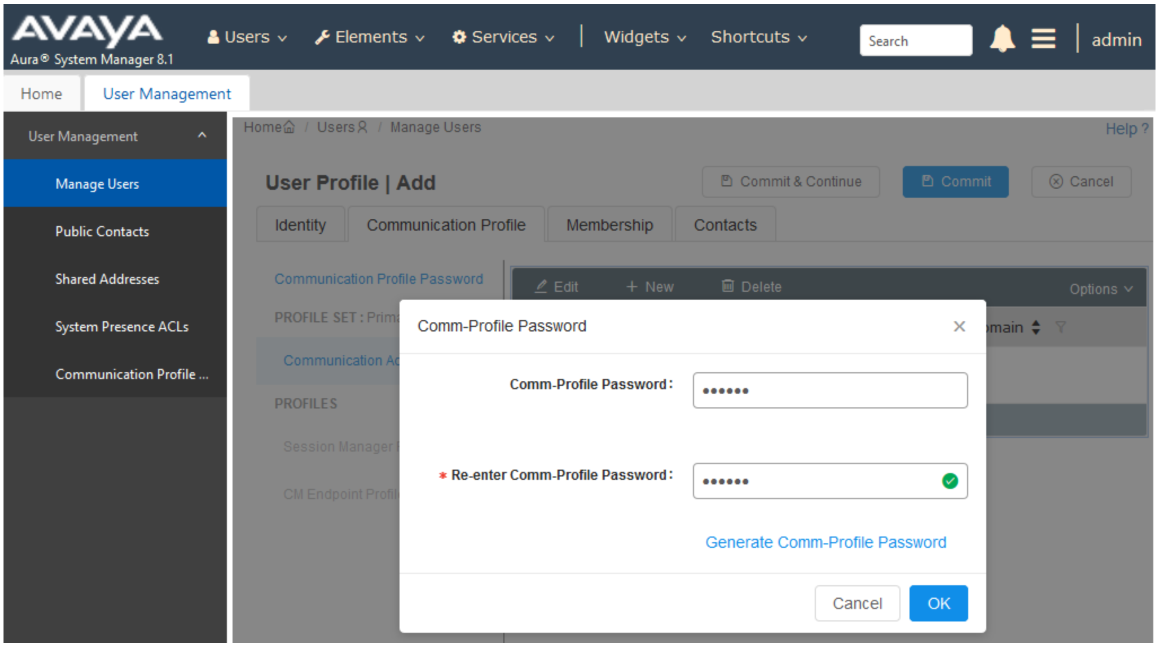
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "*<ext>@<domain>*", where "*<ext>*" is the desired ControlSpace EX-1280C SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.



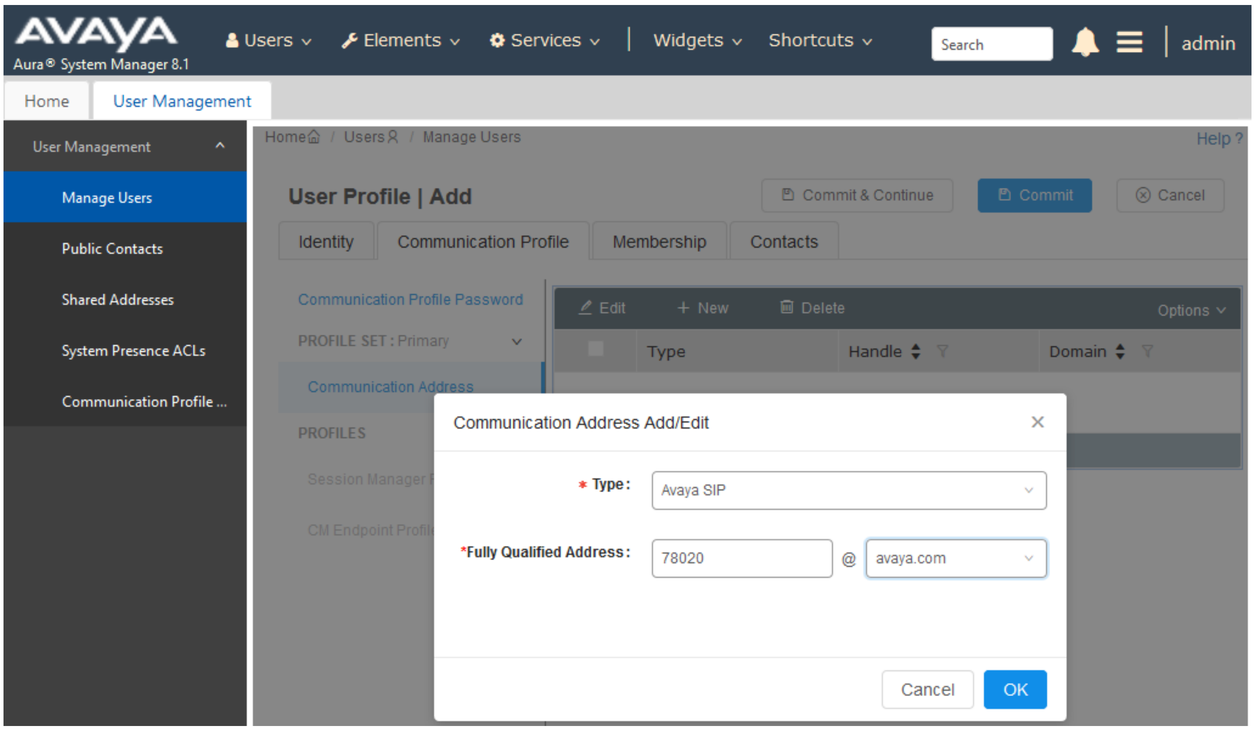
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



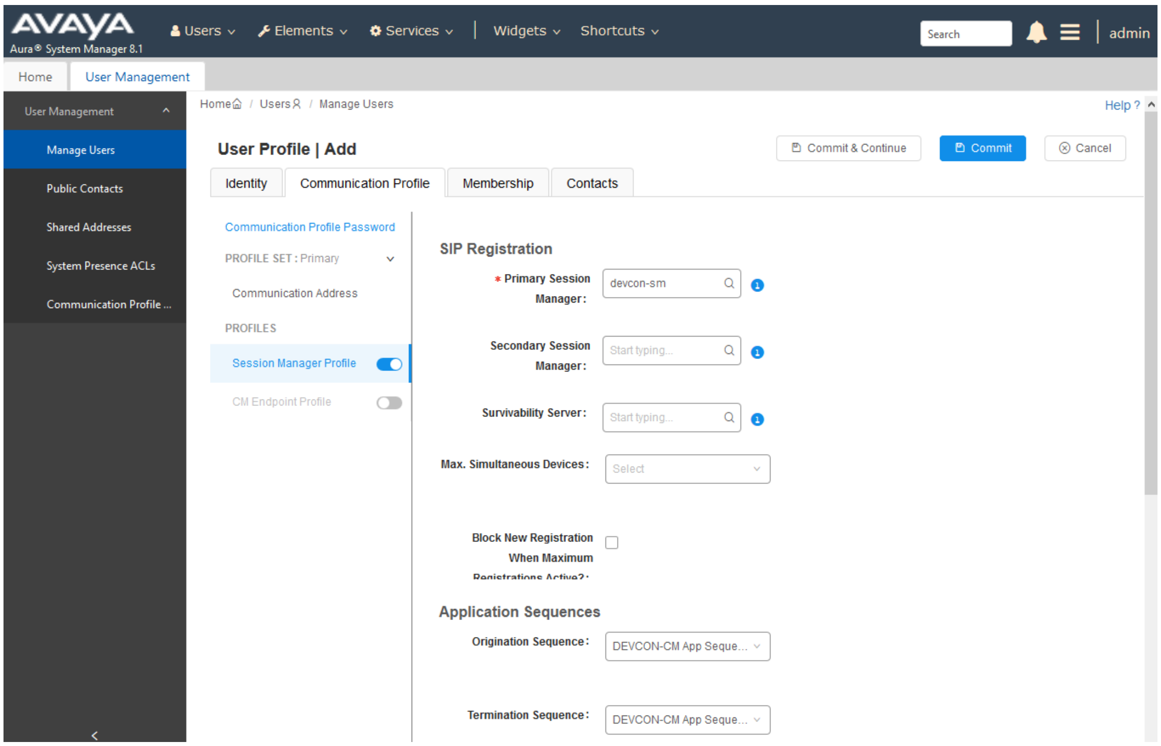
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**. Click **OK**.

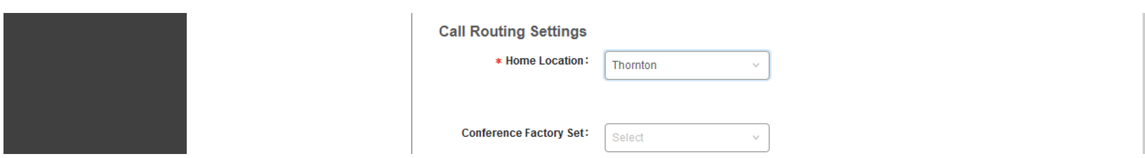


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

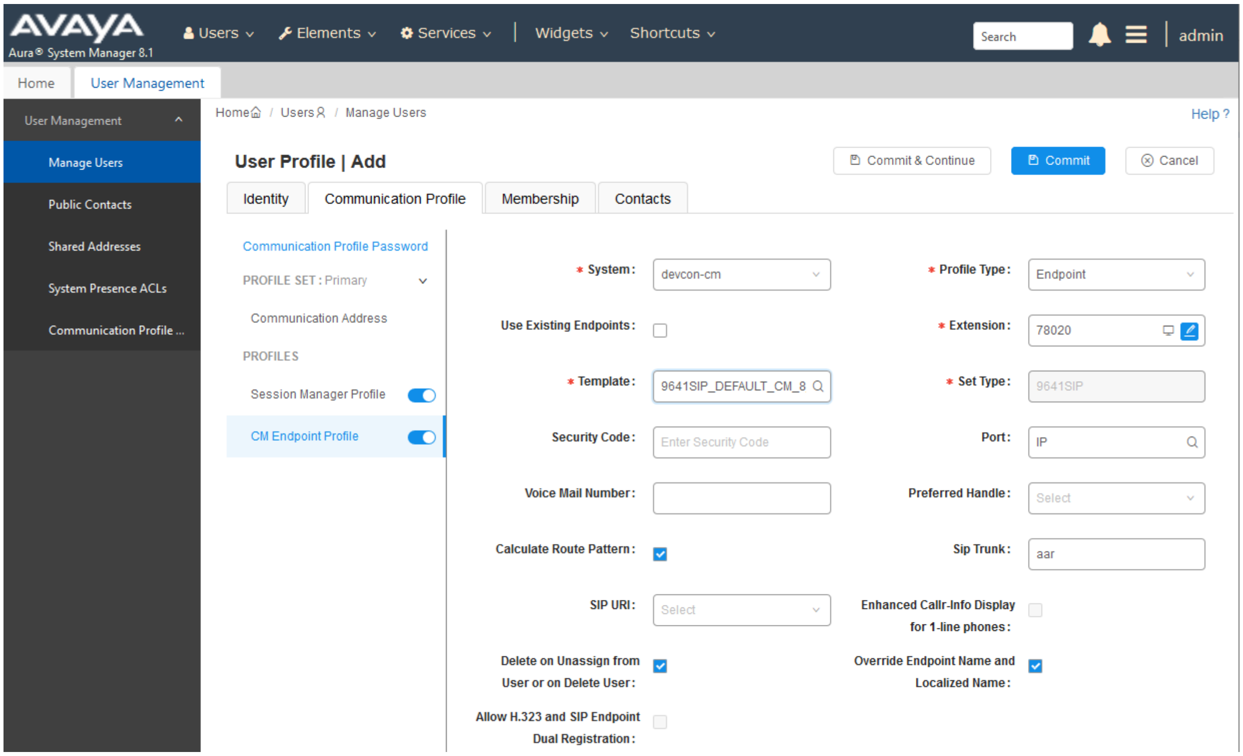


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

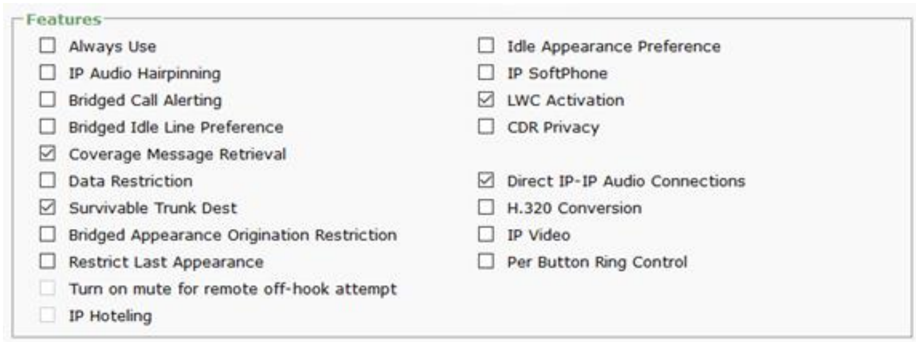


6.3.5. CM Endpoint Profile

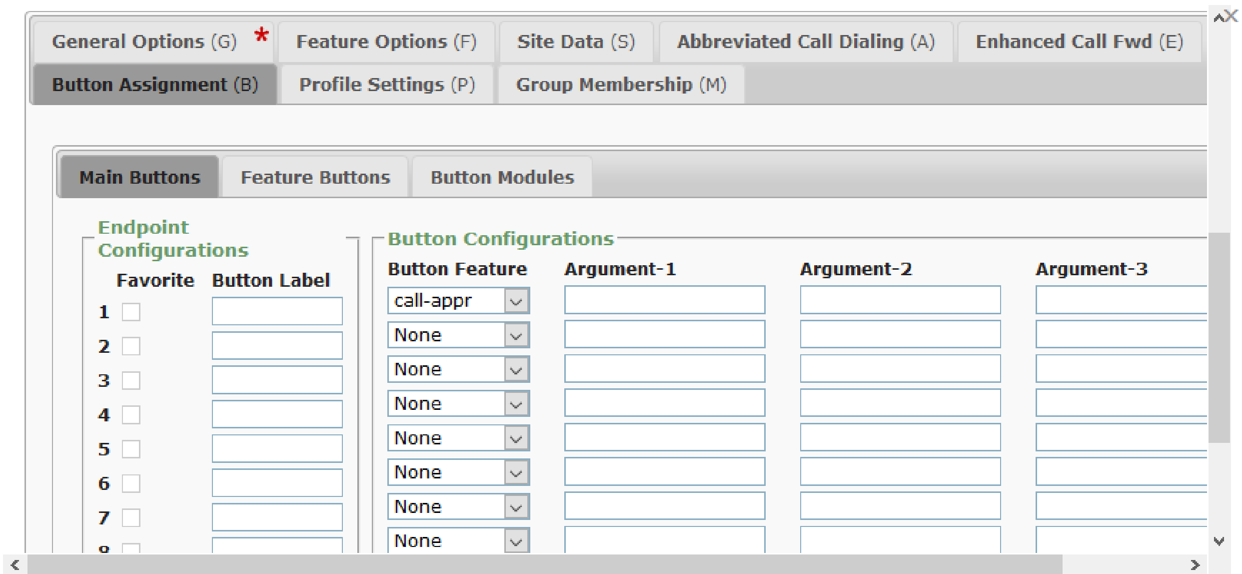
Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9651SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e, Edit icon in **Extension** field) to modify the number of call appearances supported.



Navigate to the **Feature Options** tab and scroll down to the **Features** section and disable **Restrict Last Appearance**.



In the **Button Assignment** tab, allow only one call appearance for the ControlSpace EX-1280C SIP user. Click **Done** (not shown) when complete, followed by **Commit** on the previous page.

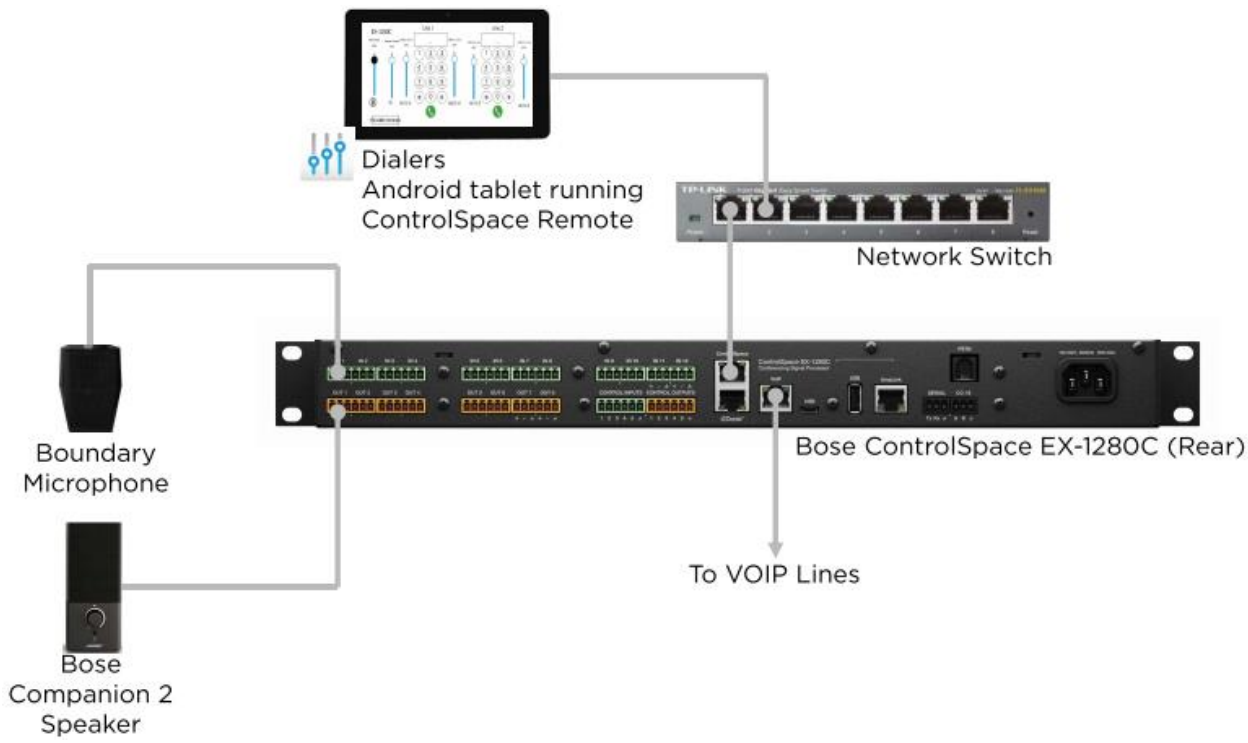


7. Configure Bose ControlSpace EX-1280C Conferencing Processor

This section covers the configuration of the Bose ControlSpace EX-1280C Conferencing Processor. The following procedures are covered:

1. ControlSpace Designer Configuration
2. Launch the Web Admin Interface
3. Accounts Configuration
4. Audio Configuration
5. Enable Auto-Answer, if desired

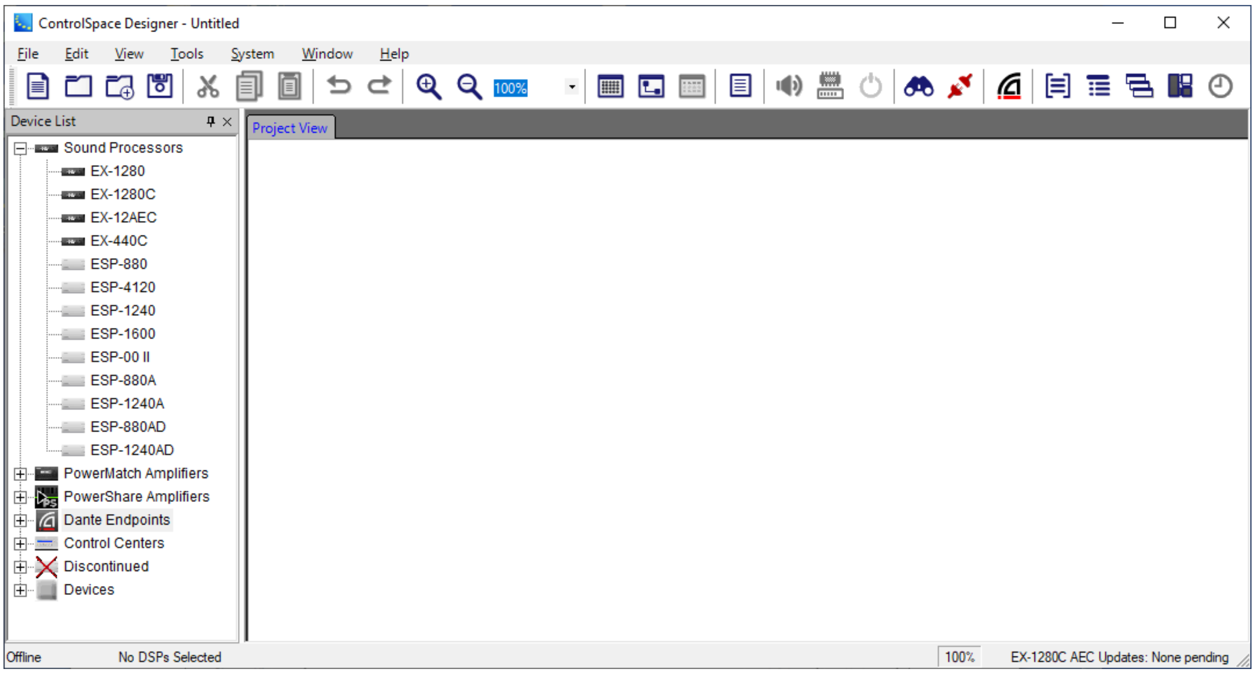
The diagram below shows the physical connectivity of the ControlSpace EX-1280C with the ControlSpace Remote, microphone, and speaker. The PC running ControlSpace Designer would also be connected to the private network switch in the diagram (not shown). The ControlSpace port at the rear of the ControlSpace EX-1280C connected to the private network switch and the VoIP port connected to the Avaya SIP network used to register with Session Manager.



7.1. ControlSpace Designer Configuration

The ControlSpace Designer (CSD) is used to control and configure the ControlSpace EX-1280C. The first step in configuring the ControlSpace EX-1280C is to use CSD to create a **.csp** project file specifying the hardware components to be used. The project file contains configuration information, settings, and control function for the ControlSpace EX-1280C.

Install the CSD software on a PC that is located on the same network as the ControlSpace EX-1280C. From the PC where CSD is installed, launch CSD to display the main window shown below.

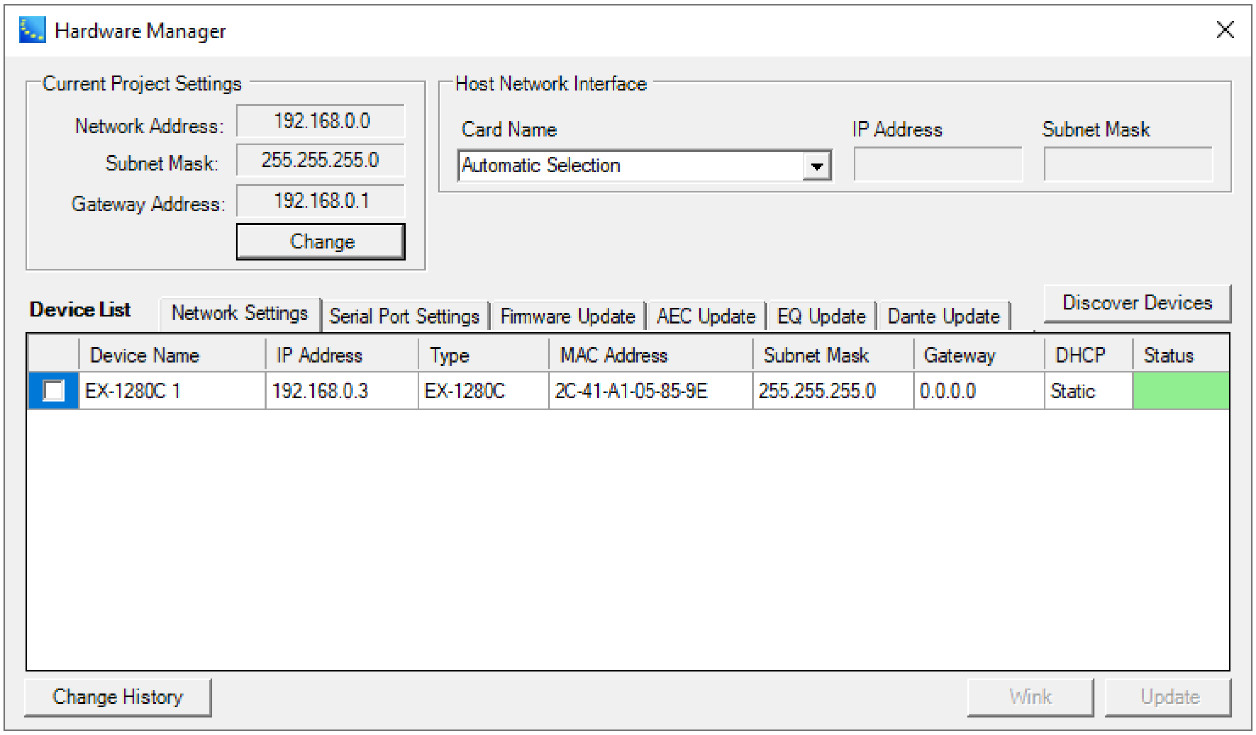


From the menu, navigate to **System → Hardware Manager**. The **Hardware Manager** window is displayed as shown below.

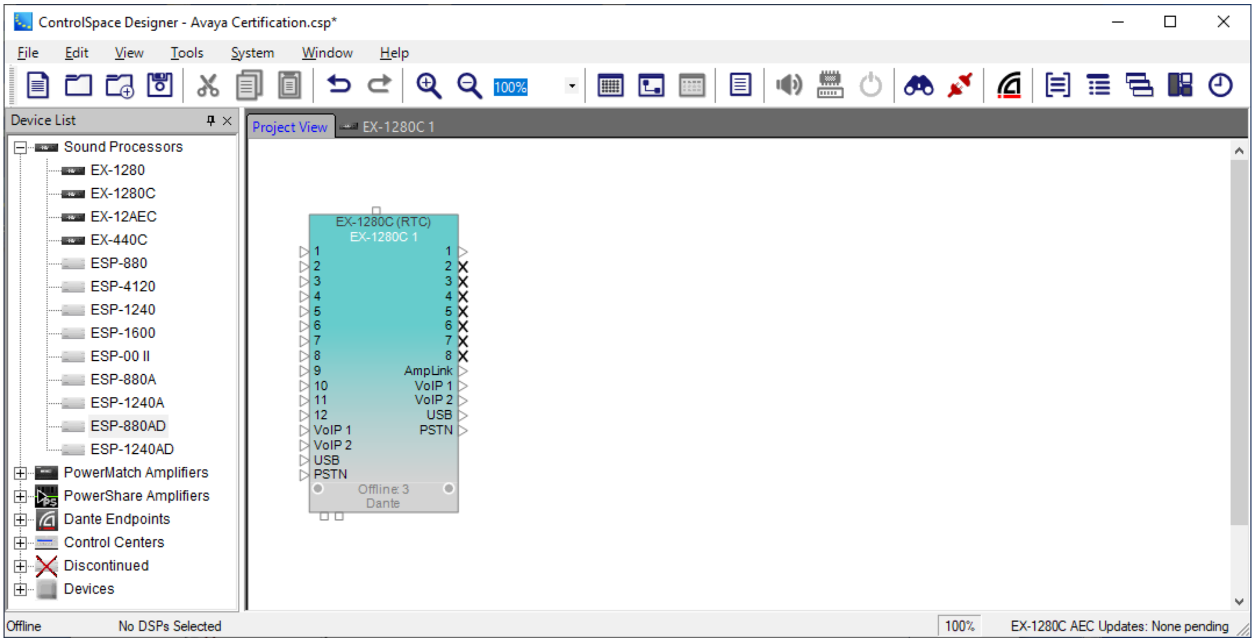
In the **Current Project Settings** section, ensure that the correct IP network settings are displayed. If not, click the **Change** button to modify as needed.

In the **Host Network Interface** section, select the appropriate network interface to be used to connect to the ControlSpace EX-1280C. In this case, **Automatic Selection** was chosen.

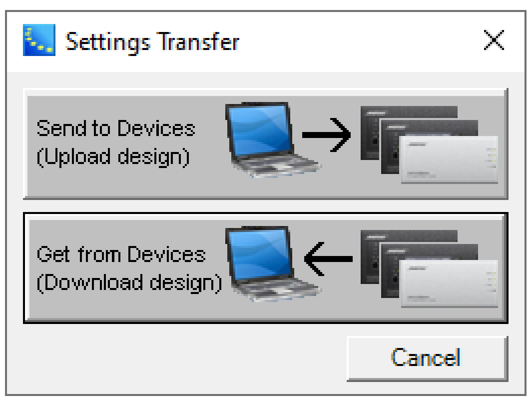
With the correct settings, the ControlSpace EX-1280C should be automatically detected and listed in the **Device List** section as shown below.



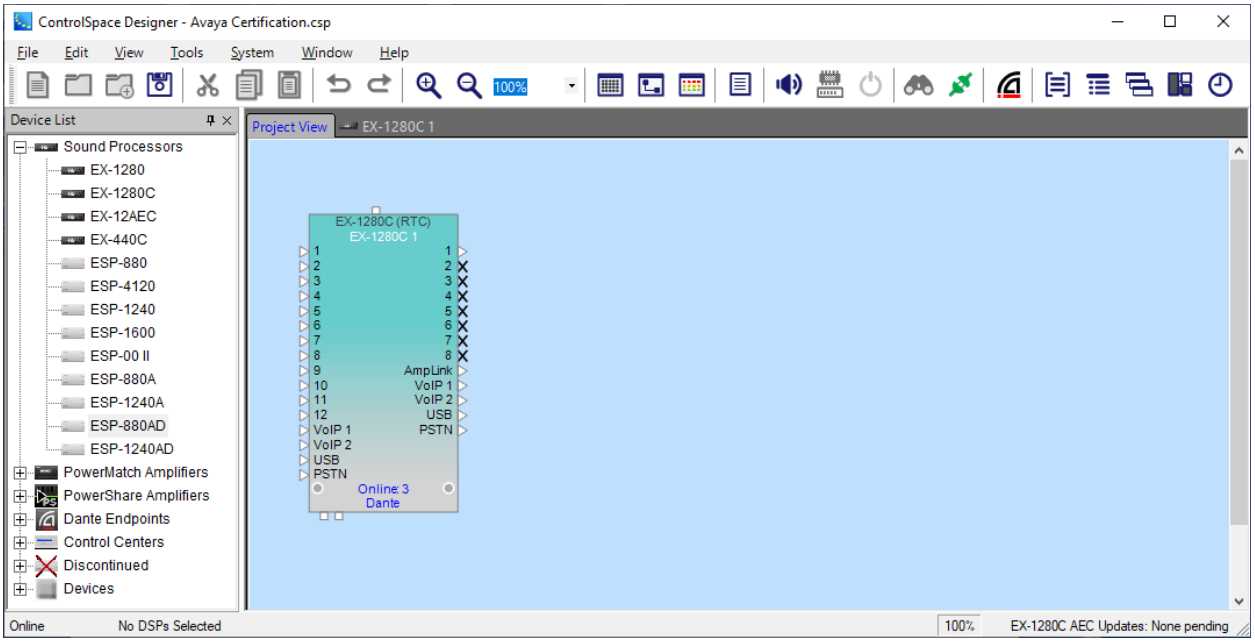
Open the **.csp** project file by navigating to **File → Open** (not shown). The project file is opened as shown below. In this case, the filename was **AvayaCertification.csp**.



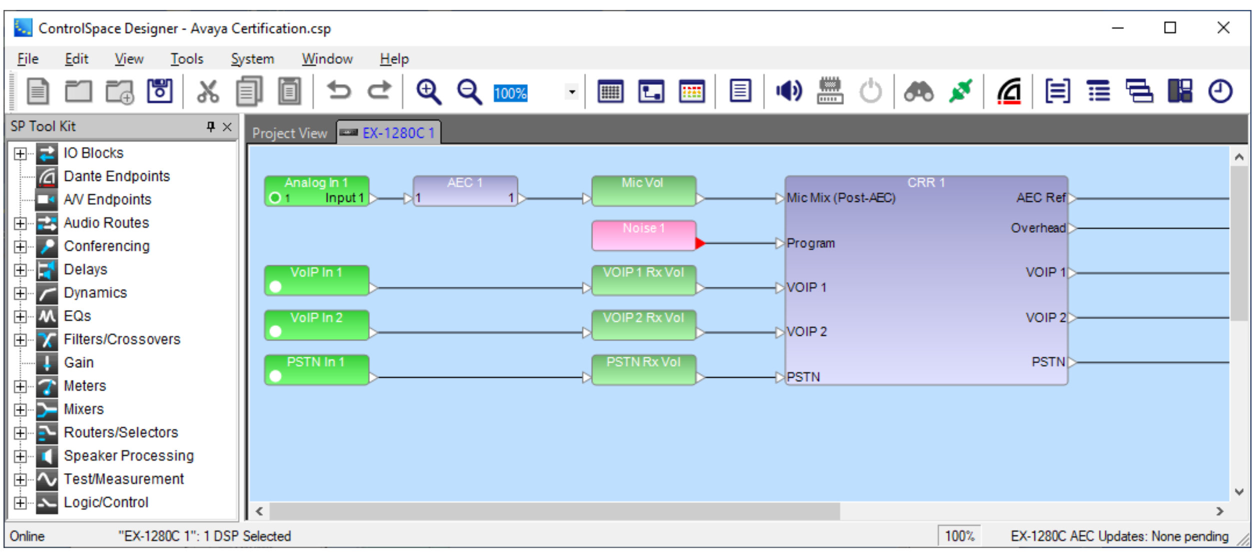
On the main menu of CSD, click on **Go Online** icon  and click on **Send to Devices (Upload design)** from the **Setting Transfer** window shown below to send the configuration and connect to the ControlSpace EX-1280C.



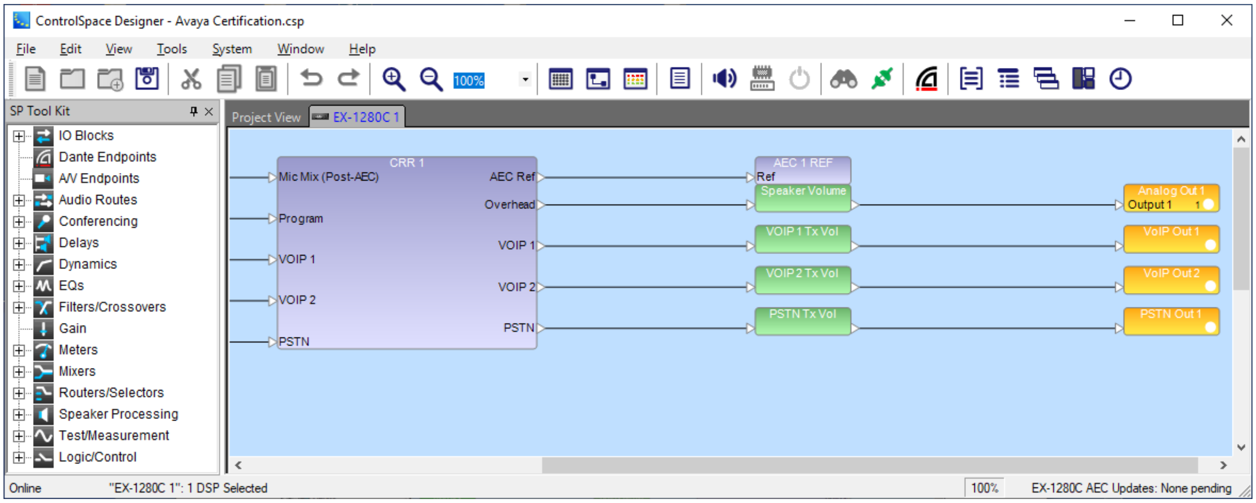
The CSD is now connected to the ControlSpace EX-1280C and the **Online** status is shown in the lower left-hand corner of the window. In addition, the window background changes to light blue.



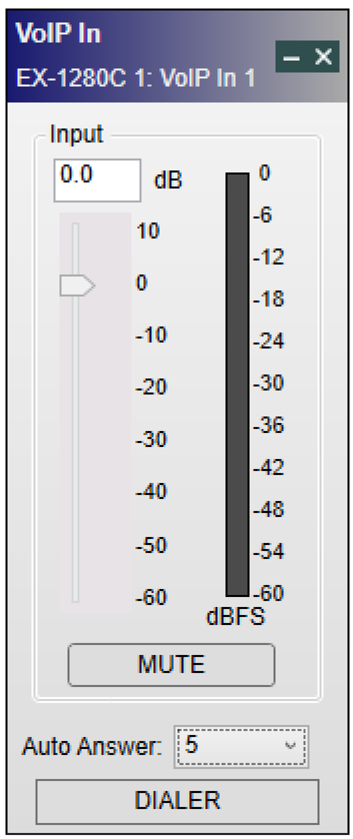
Double-click on the ControlSpace EX-1280C above to view the system components shown below.



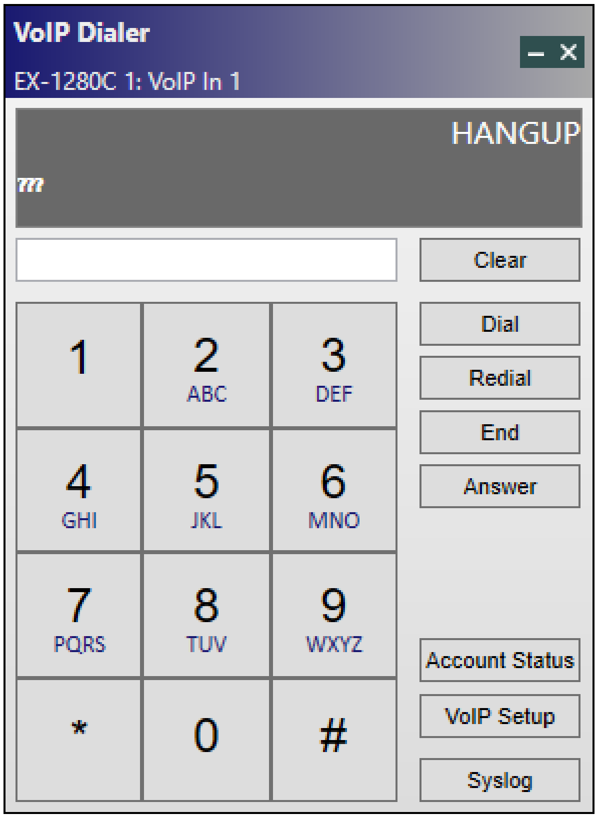
Scroll to the right, as needed, to view the rest of the system components.



Double-click on the **VoIP In 1** component to bring up the following window associated with SIP line 1. The number of rings before auto-answer may be configured here, if auto-answer will be enabled in the web admin interface. In this example, the number of rings was set to *5*.

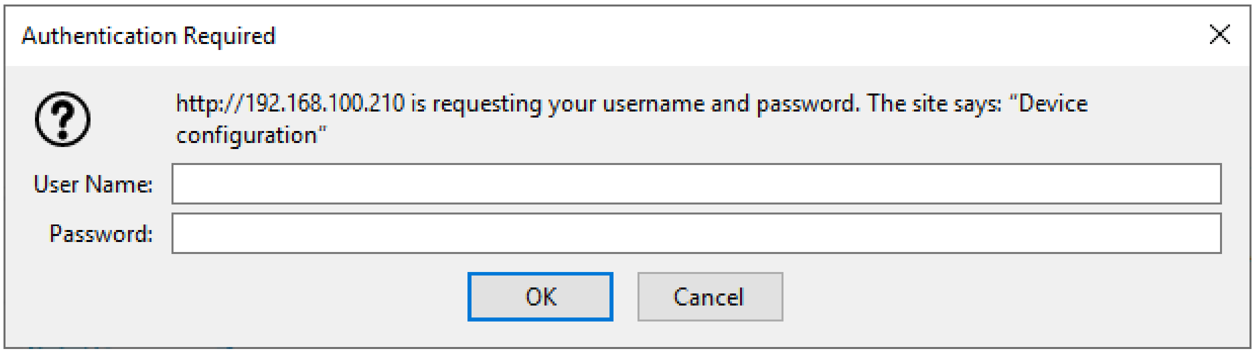


Click on the **DIALER** button above to bring up the **VoIP Dialer** as shown below for SIP line 1.



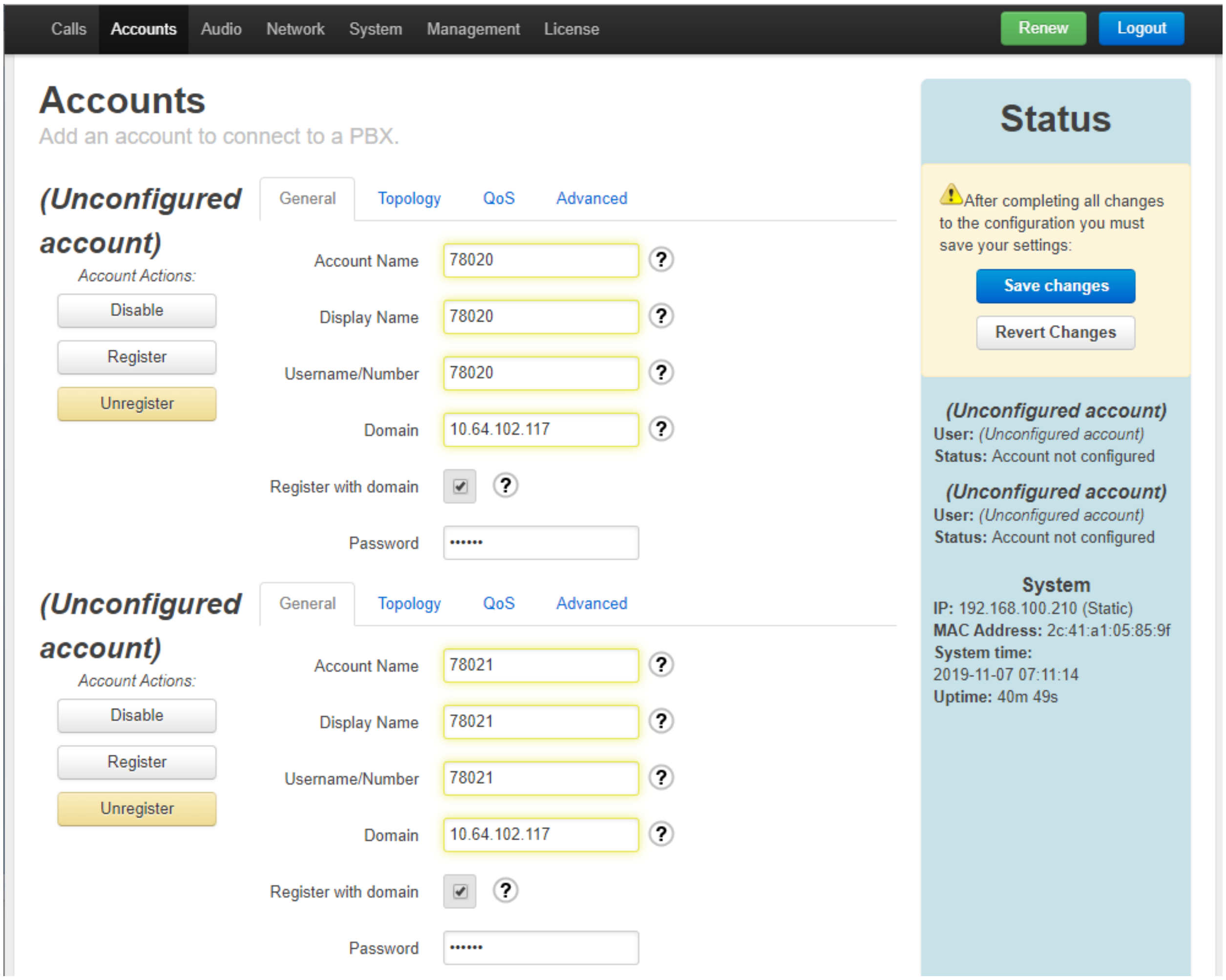
7.2. Launch the Web Admin Interface

The VoIP IP address of ControlSpace EX-1280C can be configured manually by using the rotary knob on the system front panel. Refer to **[3]** for more details. To launch the web admin interface, enter the IP address associated with the VoIP interface in a web browser and log in with the appropriate credentials.

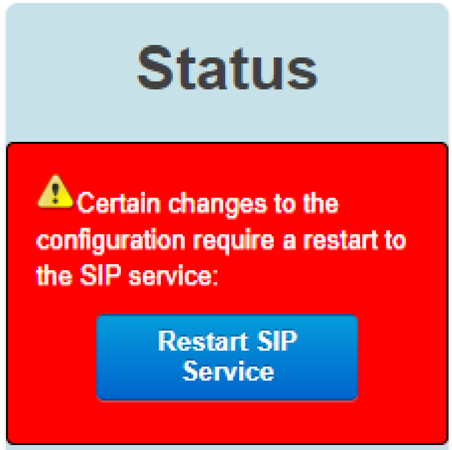


7.3. Accounts Configuration

To modify the **Accounts** configuration of the ControlSpace EX-1280C, navigate to the **Accounts** page. There are two accounts (or two SIP lines). Enter the data for SIP extensions *78020* and *78021* configured in **Section 6.3** in each account section as shown below. The **Domain** field is set to the signaling IP address of Session Manager and **Register with domain** option is enabled. Click the **Save changes** button on the right.

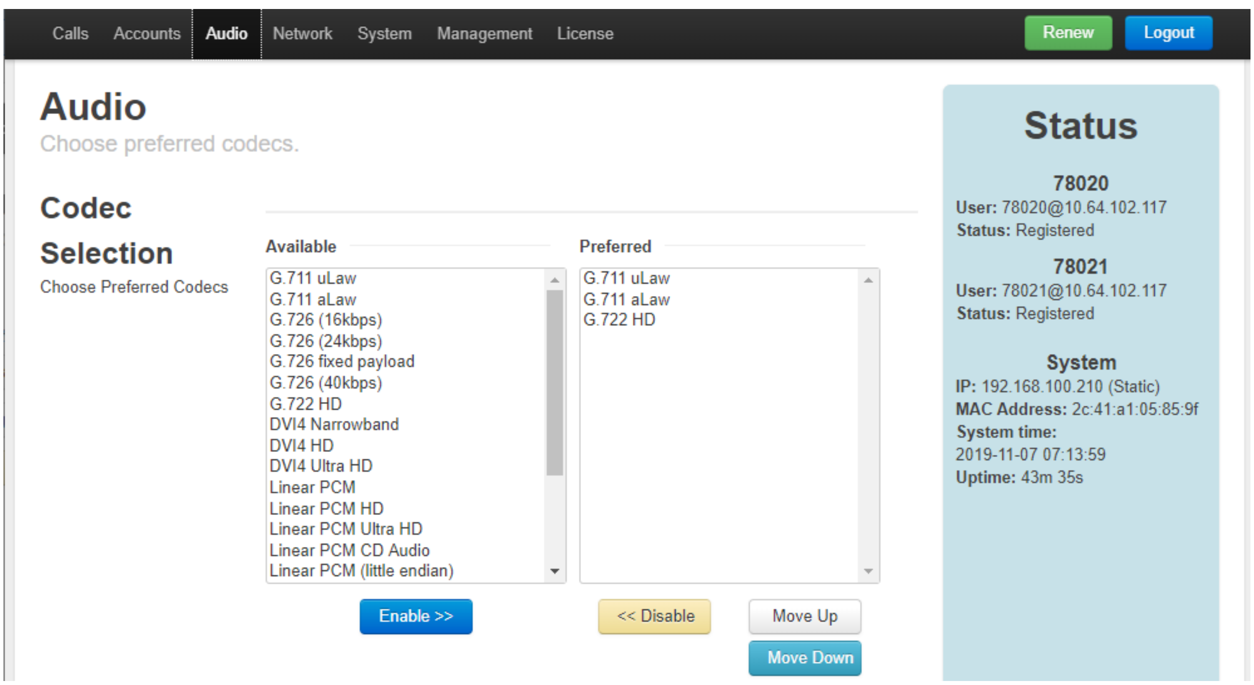


After saving the changes, click the **Restart SIP Service** button.



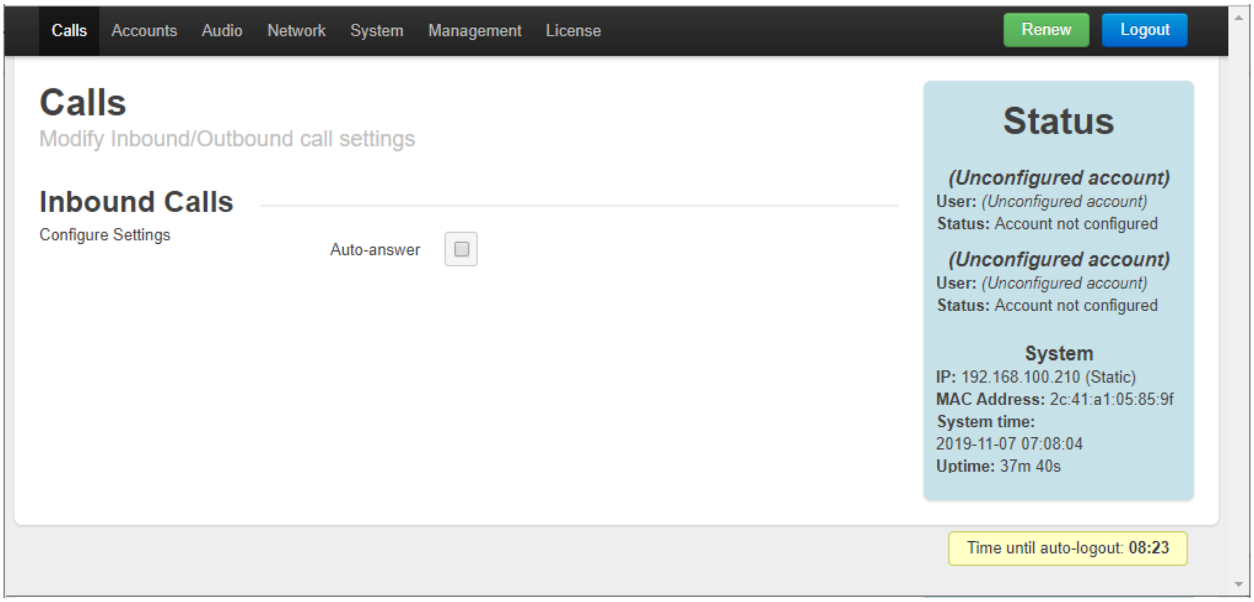
7.4. Audio Configuration

Navigate to **Audio** to configure the audio setting of the ControlSpace EX-1280C. The selected codecs are moved to **Preferred** column by selecting the available codec in the **Available** column and clicking on the **Enable >>** button. In the compliance test, the codec G.711uLaw is selected as the first choice.



7.5. Enable Auto-Answer

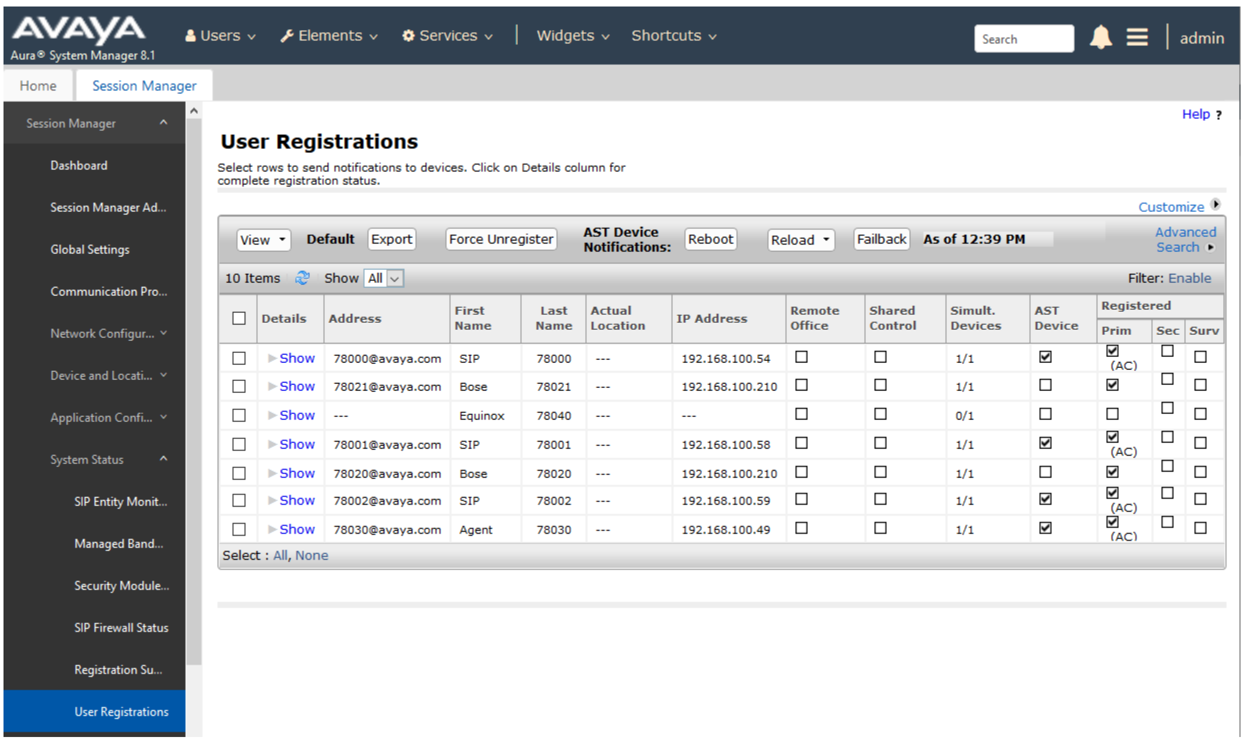
Enable **Auto-Answer,** if desired, in the **Accounts** page.



8. Verification Steps

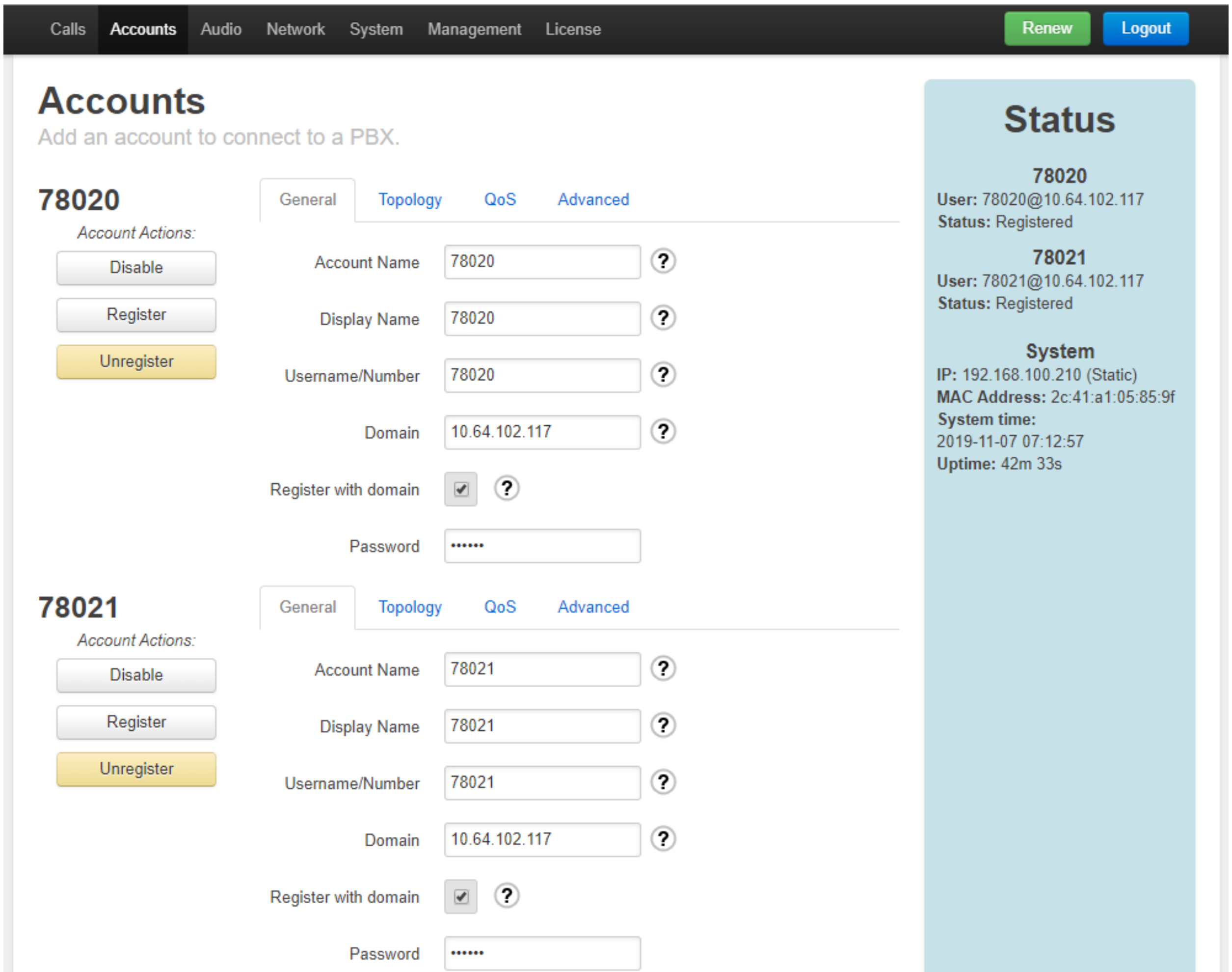
This section provides the tests that can be performed to verify proper configuration of the Bose ControlSpace EX-1280C Conferencing Processor with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the SIP lines of the ControlSpace EX-1280C have successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status as shown below.



	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered Prim	Registered Sec	Registered Surv
☐	Show	78000@avaya.com	SIP	78000	---	192.168.100.54	☐	☐	1/1	☑	☑ (AC)	☐	☐
☐	Show	78021@avaya.com	Bose	78021	---	192.168.100.210	☐	☐	1/1	☐	☑	☐	☐
☐	Show	---	Equinox	78040	---	---	☐	☐	0/1	☐	☐	☐	☐
☐	Show	78001@avaya.com	SIP	78001	---	192.168.100.58	☐	☐	1/1	☑	☑ (AC)	☐	☐
☐	Show	78020@avaya.com	Bose	78020	---	192.168.100.210	☐	☐	1/1	☐	☑	☐	☐
☐	Show	78002@avaya.com	SIP	78002	---	192.168.100.59	☐	☐	1/1	☑	☑ (AC)	☐	☐
☐	Show	78030@avaya.com	Agent	78030	---	192.168.100.49	☐	☐	1/1	☑	☑ (AC)	☐	☐

2. Alternatively, the ControlSpace EX-1280C web admin interface also provides the SIP line status on the right as shown below.



3. Verify basic telephony features by establishing calls between the ControlSpace EX-1280C and another Avaya IP phone.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Bose ControlSpace EX-1280C Conferencing Processor with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The ControlSpace EX-1280C successfully registered with Session Manager and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2.**

10. Additional References

This section references the Avaya and Bose documentation relevant to these Application Notes.

The following Avaya product documentation is available at support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 2, July 2019.
[2] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019.

The following Bose documentation available from Bose.

[3] *Bose ControlSpace Installation & Operation Guide.*

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 DevConnect Program at devconnect@avaya.com.

Bose Professional

To: Avaya DevConnect

DATE
09 27 2019

SUBJECT
Declaration of conformance for Bose ControlSpace EX platform

We, Bose Corporation, hereby confirm that the following ControlSpace conferencing processors:

EX-1280C

EX-440C

are based on the same platform and therefore:

Use an identical SIP stack

Use identical interface components

Use the same firmware on the VOIP interface as of version 1.930 build 1

Use the same configuration software as of ControlSpace Designer 5.5

The difference in the two conferencing processors is their input-output configurations and the number of VOIP lines that are supported:

EX-1280C has two lines

EX-440C has just one

Best regards,

Martin Bodley
Director, Emerging Business

Bose Corporation
Corporate Office

The Mountain
Framingham, MA 01701-9168
US

+1 508 879 7330

bose.com