Avaya Solution & Interoperability Test Lab

Application Notes for CSS Mindshare 100500 MaxPlus Dispatch Console integration with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate CSS Mindshare 100500 MaxPlus Dispatch Console 3.27.2 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. CSS Mindshare 100500 MaxPlus Dispatch Console incorporates telephony to integrate both radio and telephone functions. This solution also includes Console Builder for creating a user console.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 CSS Mindshare 100500 MaxPlus Dispatch Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. CSS Mindshare 100500 MaxPlus Dispatch Console is purposely built for radio dispatch applications required for 24/7 operation. By integrating PC and audio processor components into a single device, CSS Mindshare 100500 Max Plus Dispatch Console provides a complete dispatch console workstation.

2. General Test Approach and Test Results

The interoperability testing scope is limited to MaxPlus Dispatch Console telephony integration with Session Manager and Communication Manager. The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between MaxPlus Dispatch Console, Avaya SIP / H.323 desk phones, and the PSTN, and exercising basic telephony features, such as hold, mute, and transfer. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs).

The serviceability testing focused on verifying that MaxPlus Dispatch Console comes back into service after IP network interruption.

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 these DevConnect Application Notes 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 these Application Notes, the interface between Avaya systems and MaxPlus Dispatch Console did not include use of any specific encryption features as requested by CSS Mindshare.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP user registration of MaxPlus Dispatch Console with Session Manager.
- Calls between MaxPlus Dispatch Console and Avaya SIP / H.323 Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between MaxPlus Dispatch Console and the PSTN.
- G.711MU and G.729 codec support.
- Proper DTMF tone generation.
- Basic telephony features including hold, mute, redial, multiple calls, and blind and attended call transfer.
- Extended telephony features using Communication Manager FACs and for Call Forward, Call Park/Unpark, and Call Pickup.
- Use of programmable buttons (Console Builder button controls) for speed dial on MaxPlus Dispatch Console.
- Proper system recovery after a loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations:

- Call Conferencing is not supported.
- Voicemail MWI is currently not supported.
- Long hold recall timer is not supported.
- Audio tones for invalid numbers or outbound call restriction are not given but Console line indicator display notifications are made.
- The Console line indicator display does not show called parties. The Console line Indicator displays the calling party only during ringing. Once that call is answered, the display clears. The display will continue to show the calling party if the call is not answered as per MaxPlus Dispatch Console design. If multiple consoles register to the same user, the display for the console(s) that did not answer the call will continue to show the calling party.
- MDA (multi-device access) with non-TLS support will not alert DUT instances if an Avaya SIP phone is registered with the same user as the console.
- There is no indication that MDA registration over the maximum allowed users is attempted. It is not allowed but the only indication is that the console buttons are inactive. Configuring the SIP user to not allow registration when the user limit is reached is recommended.

2.3. Support

For technical support and information on MaxPlus Dispatch Console, contact CSS Mindshare Technical Support at:

- Phone: +1 402-261-8688 x2
- Email: techsupport@css-mindshare.com
- Website: https://support.css-mindshare.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network:

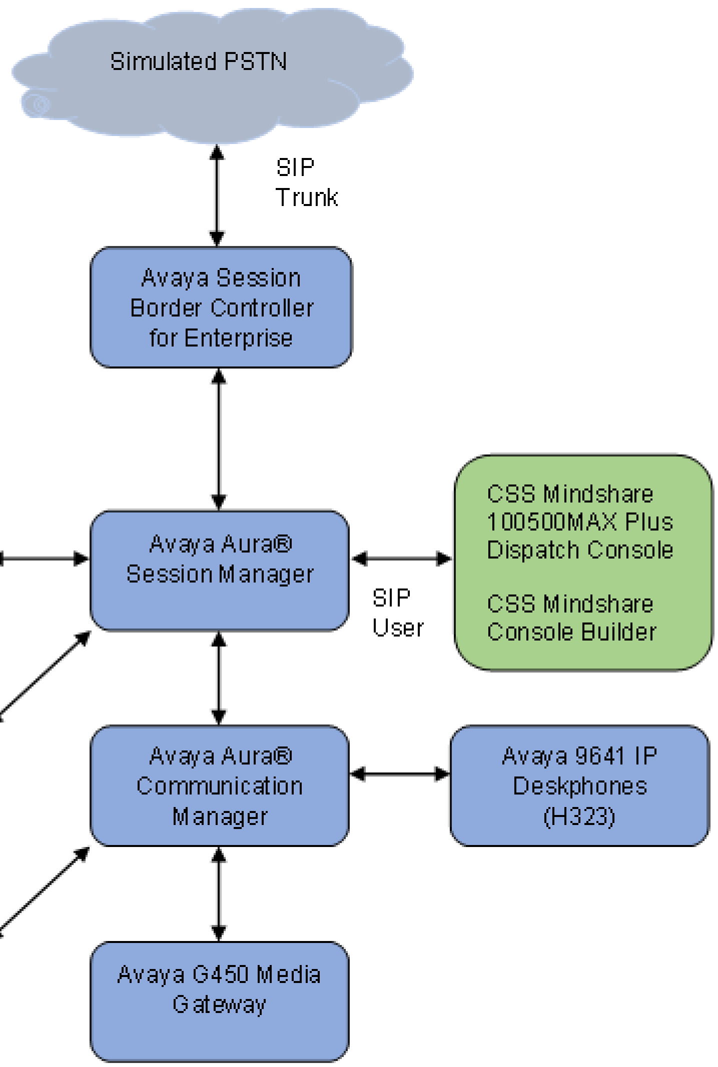


Figure 1: Avaya SIP Network with CSS Mindshare 100500 MaxPlus Dispatch Console

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 running on Virtual Machine	01.0.890.0-27168
Avaya Aura® Session Manager running on Virtual Machine	8.1.3.3_813310
Avaya Aura® System Manager running on Virtual Machine	8.1.3.3.1013529
Avaya Session Border Controller for Enterprise running on Virtual Machine	8.1.3.0-31-21052
Avaya G450 Media Gateway	41.34.1
Avaya Aura® Media Server	8.0.0.21
Avaya 9641G IP Deskphone	6.8511 (H.323)
Avaya J179 IP Phone	4.0.9.0.4 (SIP)
CSS Mindshare 100500 MaxPlus Dispatch Console	3.27.2 Debian GNU/Linux 10 (buster) Gnome 3.30.2
CSS Mindshare Console Builder	3.27.2

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 Network Region
- Administer IP Codec Set

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

Note: It is assumed that basic configuration of Communication Manager has already been completed, such as the SIP trunk to Session Manager. However, implementers should ensure sufficient Maximum Administered SIP Trunks licenses are available to accommodate the traffic between Communication Manager and Session Manager. The SIP station configuration for MaxPlus Dispatch Console is configured through System Manager in **Section 6.3**.

5.1. Verify Communication Manager license

Using the SAT, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed 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.

```
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    254
                                      Maximum Stations:   150    119
                              Maximum XMOBILE Stations: 36000      0
                    Maximum Off-PBX Telephones - EC500:   150      0
                    Maximum Off-PBX Telephones -   OPS:   150     63
                    Maximum Off-PBX Telephones - PBFMC:   150      0
                    Maximum Off-PBX Telephones - PVFMC:   150      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 Network Region

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 G450 Media Gateway or Media Server. The **ip-network-region** form also specifies the **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: Main                     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: 3329
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
```

5.3. Administer IP Codec Set

In the **ip-codec-set** form, select the audio codec type supported for calls routed over the SIP trunk to MaxPlus Dispatch Console. Note that IP codec set **1** was specified in IP Network Region **1** shown above. The settings of the **ip-codec-set** form are shown below. Ensure **none** is one of the **Media Encryption** options offered. MaxPlus Dispatch Console was tested using G.711MU and G.729 codecs.

```
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: G.729             n         2        20
 3:
 4:
 5:
 6:
 7:

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

6. Configure Avaya Aura® Session Manager

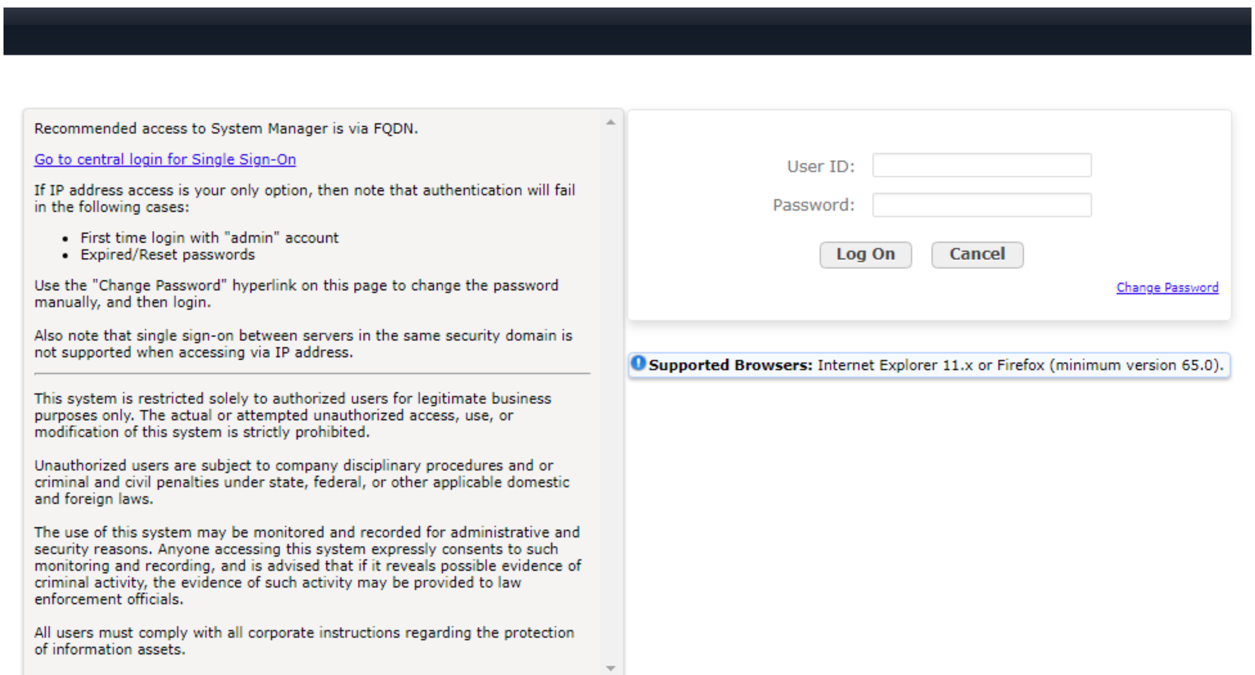
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol
- Administer SIP User

Note: It is assumed that Session Manager and Communication Manager SIP trunk connections are configured. This section will focus on the configuration of a SIP user for MaxPlus Dispatch Console.

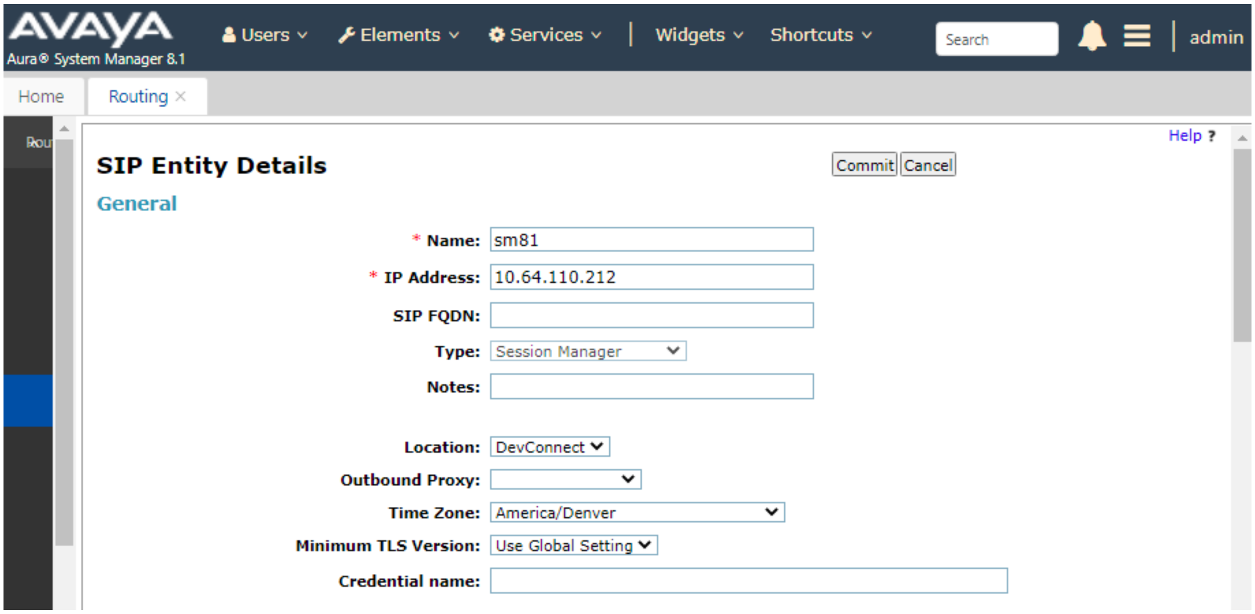
6.1. Launch System Manager

Access System Manager web interface by entering **http://<ip-address>/SMGR** in a web browser, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials.

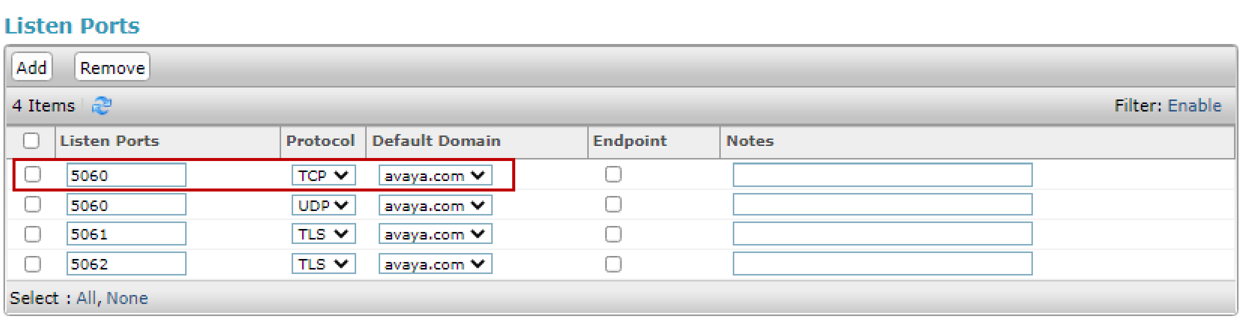


6.2. Set Network Transport Protocol

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 MaxPlus Dispatch Console is specified in the list below. For the compliance test, MaxPlus Dispatch Console used **TCP** network transport.



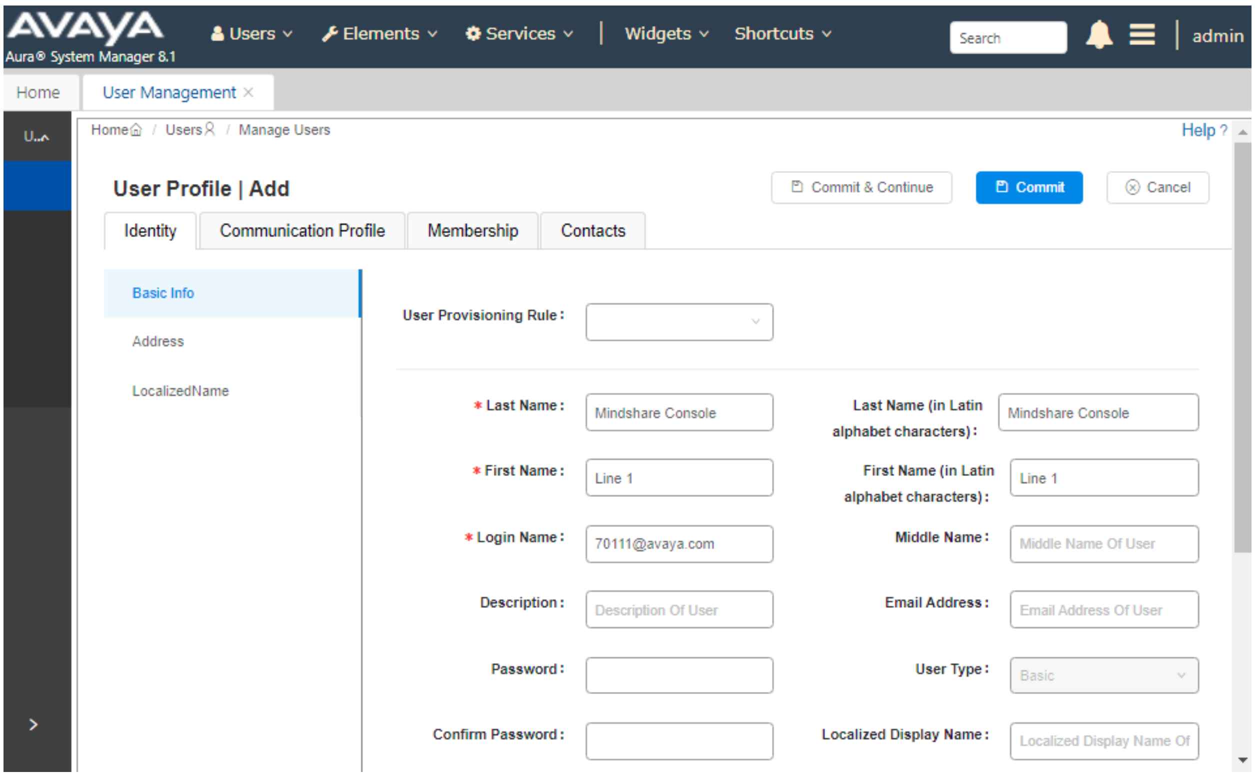
6.3. Administer SIP Users

A SIP user must be configured for MaxPlus Dispatch Console by the following steps. This configuration is automatically synchronized with Communication Manager. In Session Manager, select **Users →User Management →Manage Users** to display the **User Management** screen (not shown). Click **+ New** to add a user.

6.3.1. Identity

Enter values for the following required attributes for a new SIP user in the **User Profile** screen:

- **Last Name**: Enter the last name of the user, e.g., **Mindshare Console**.
- **First Name**: Enter the first name of the user, e.g., **Line 1**.
- **Login Name**: Enter <extension>@<sip domain> of the user (e.g., **70111@avaya.com**).

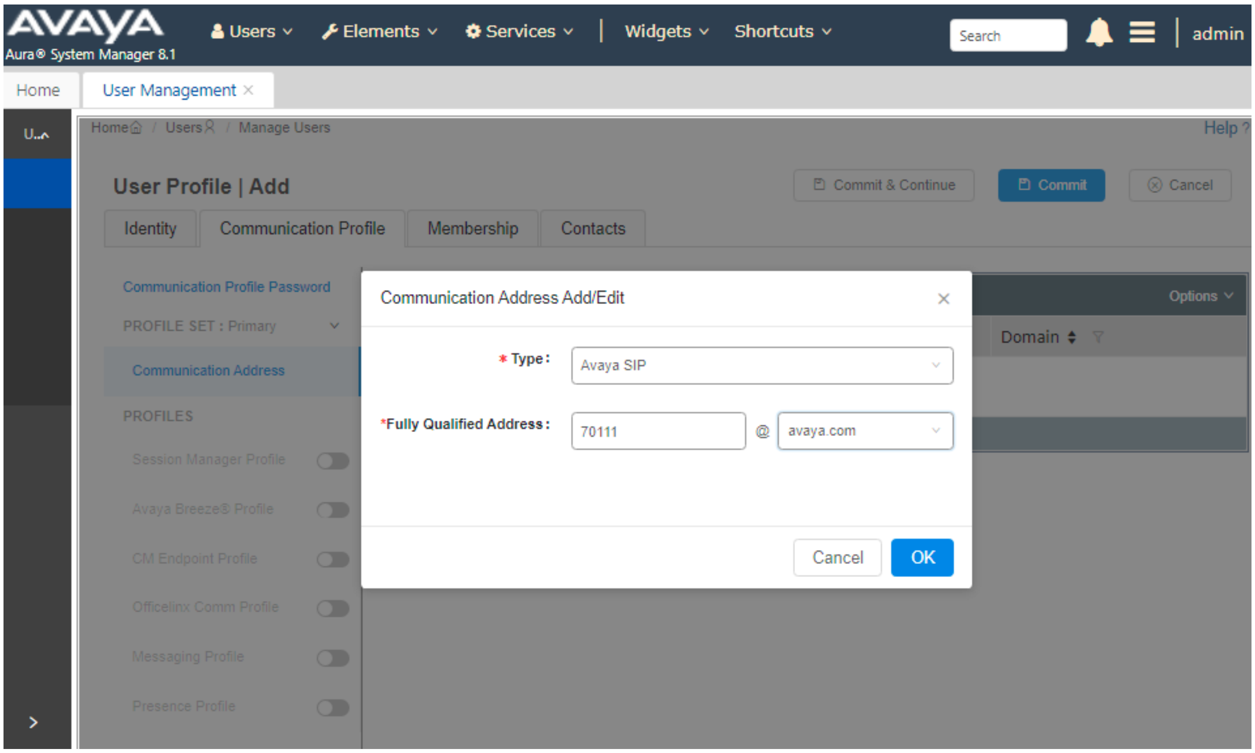


6.3.2. Communication Address

Select the **Communication Profile** tab. Select **Communication Address** in the left list and click **+ New** (not shown).

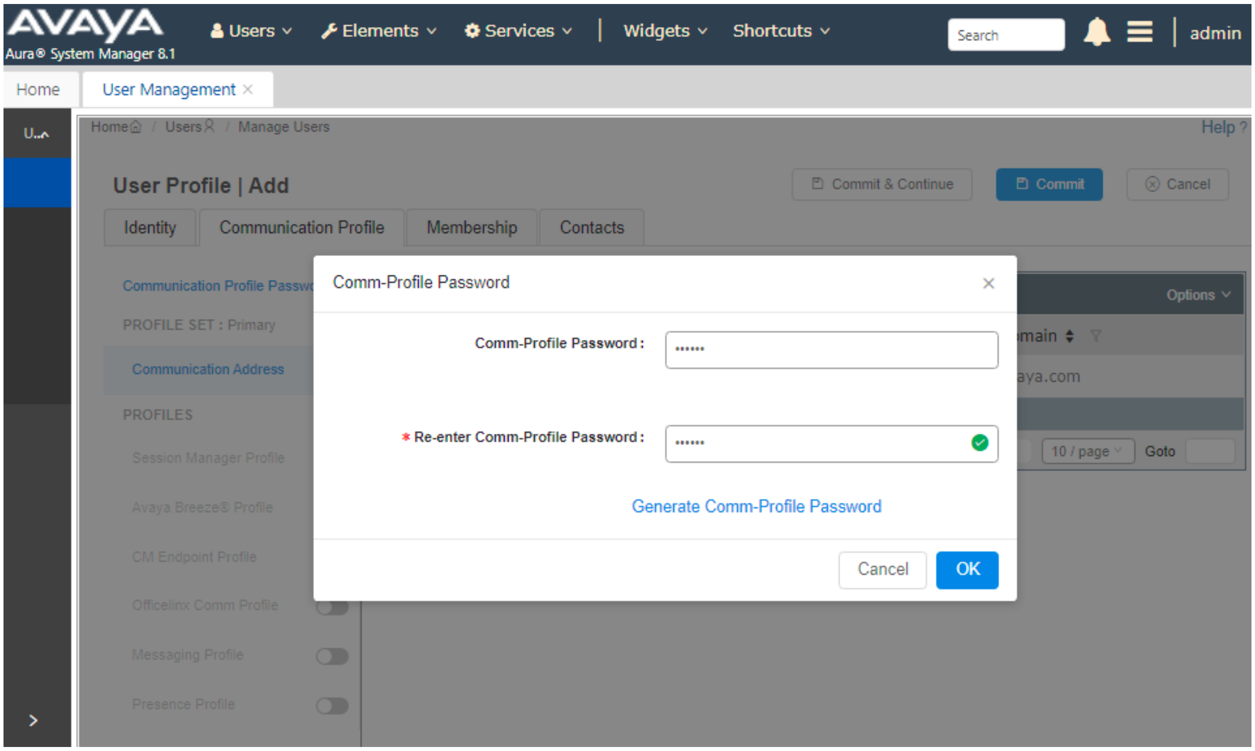
Enter the following attributes for the **Communication Address**:

- **Type:** Select **Avaya SIP** from the drop-down list.
- **Fully Qualified Address:** Enter the extension number (e.g.,**70111**).
- **Domain:** Enter the domain (e.g., **avaya.com**).



6.3.3. Communication Profile Password

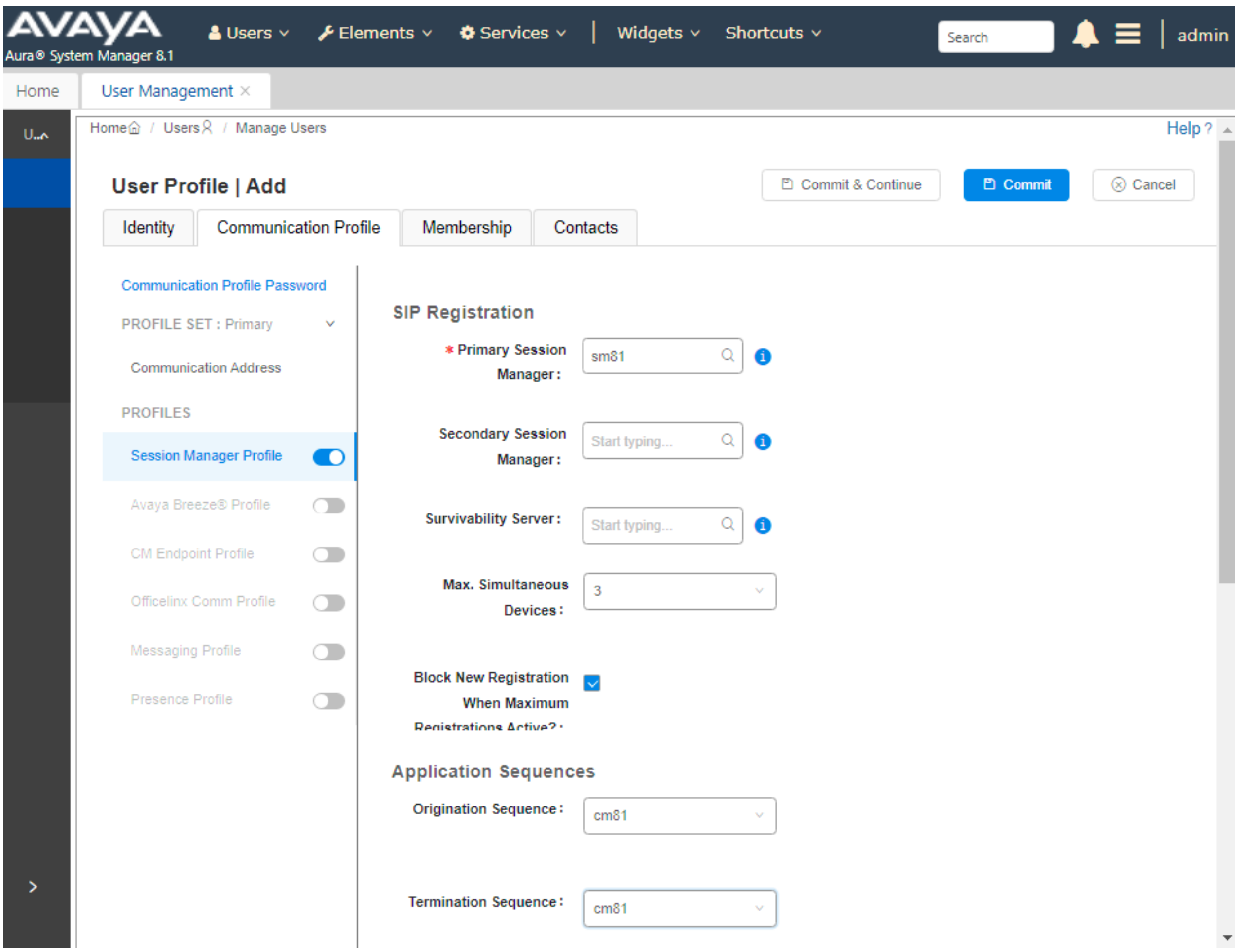
Select **Communication Profile Password** on the left and in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields, enter a password. This will be used to register the device. Click **OK**.



6.3.4. Session Manager Profile

Click on the **Session Manager Profile** slide button. For **Primary Session Manager**, **Origination Sequence**, **Termination Sequence**, and **Home Location** (not shown), select the values corresponding to the applicable Session Manager and Communication Manager. Select **3** for **Max. Simultaneous** Devices. Check **Block New Registration When Maximum Registrations Active?** to ensure any new console registration requests when the maximum concurrent registrations are being used will not drop a current registration to accept the new request. Retain the default values in the remaining fields.

Note: The maximum supported number of simultaneous registrations with this user (MDA) is **10**. MaxPlus Dispatch Console interoperability testing used **3**.

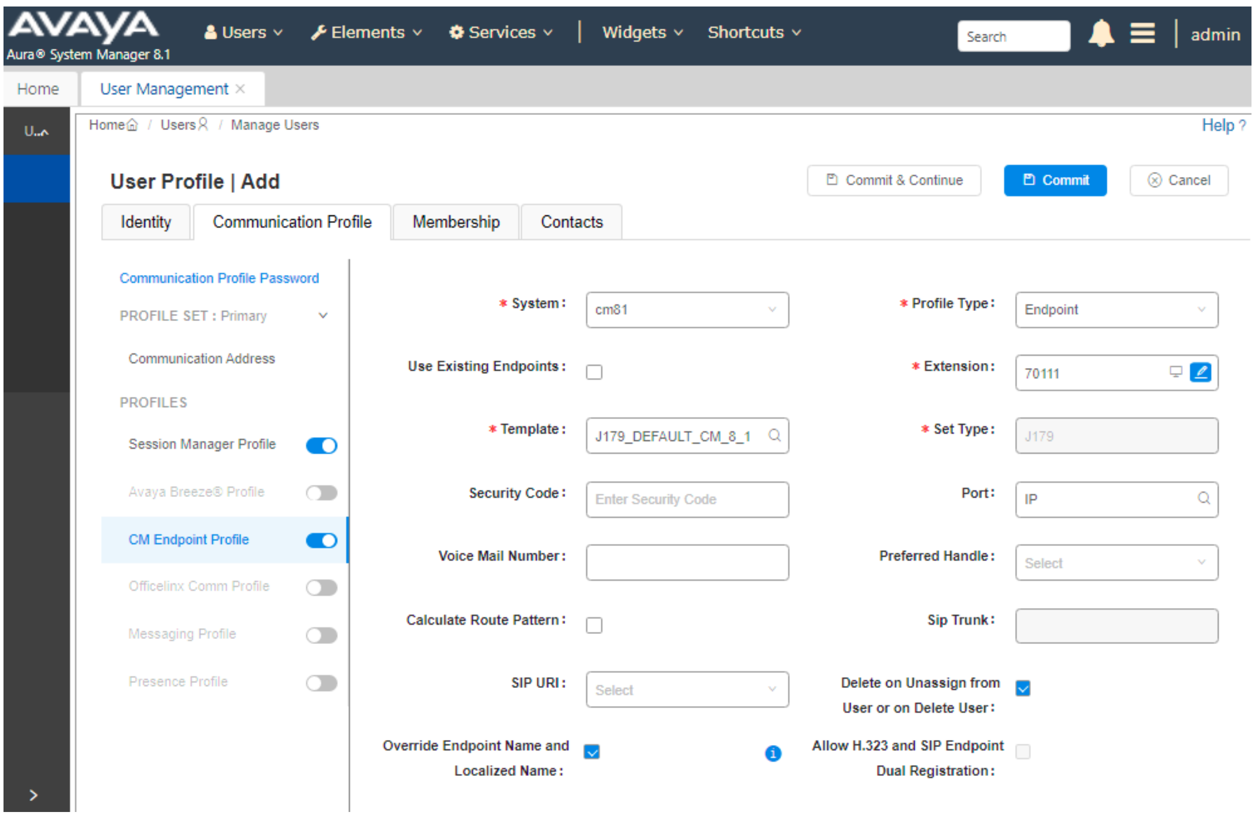


6.3.5. CM Endpoint Profile

Click on the **CM Endpoint Profile** slide button. Fill in the following fields:

- **System**: Select the relevant Communication Manager SIP Entity (e.g., **cm81**).
- **Profile Type**: Select **Endpoint**.
- **Template**: Select **J179_DEFAULT_CM_8_1**.
- **Extension**: Enter the extension number (e.g., **70111**).

Click on **Endpoint Editor** in the **Extension** field to edit Communication Manager settings. Input the appropriate **Coverage Path** (not shown) configured to route unanswered calls to voicemail. Click **Done** to close the Endpoint Editor. Click **Commit** (not shown).



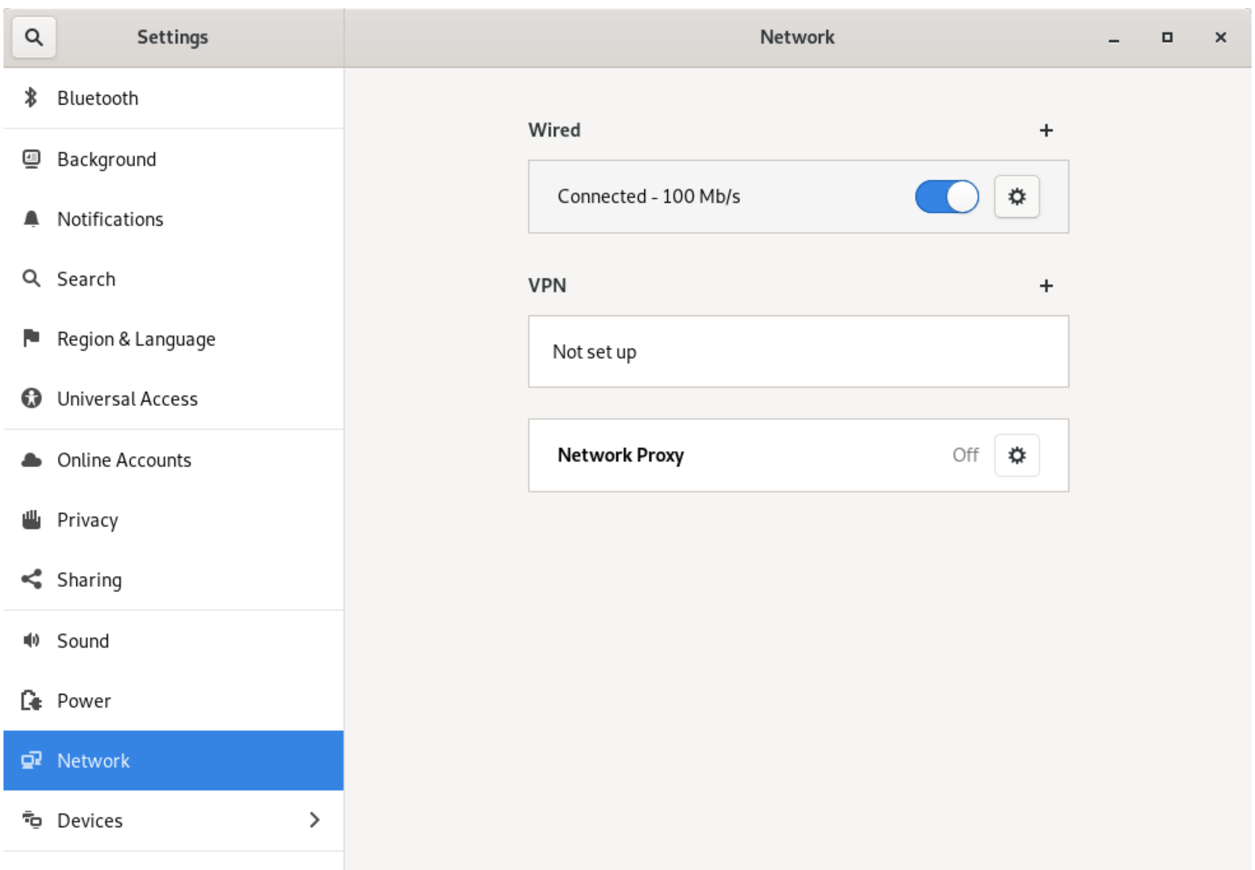
7. Configure CSS Mindshare 100500 MaxPlus Dispatch Console

This section covers MaxPlus Dispatch Console configuration using Console Builder . The procedure covers the following areas:

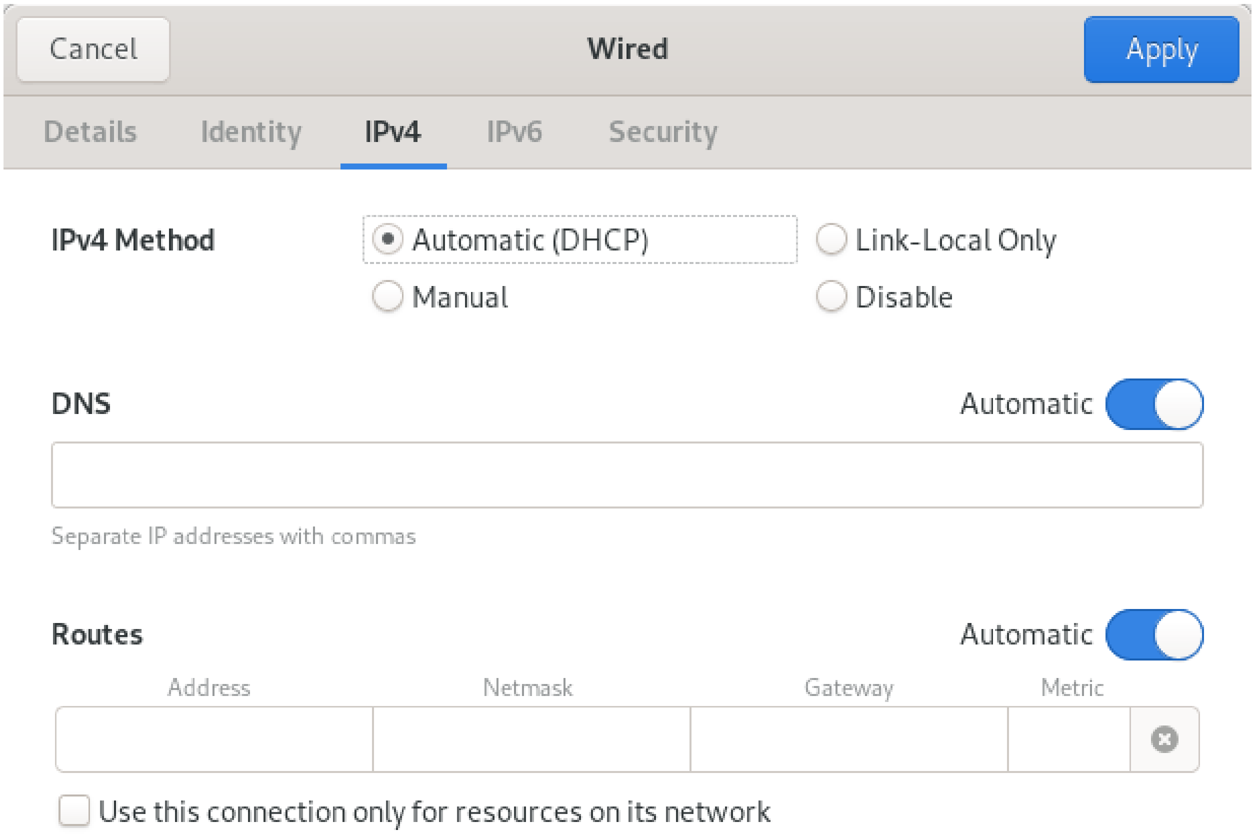
- Configure IP Address
- Launch Console Builder
- Configure Phone System Parameters
- Administer IP Comms
- Create Console Layout
- Save Layout to Configuration File

7.1. Configure IP Address

MaxPlus Dispatch Console is configured for DHCP on power up. A static IP address can be assigned via the operating system desktop. Select the **Settings** button from the **System Menu** to open the Settings dialog. Click **Network** on the left side and click the settings icon under **Wired**.

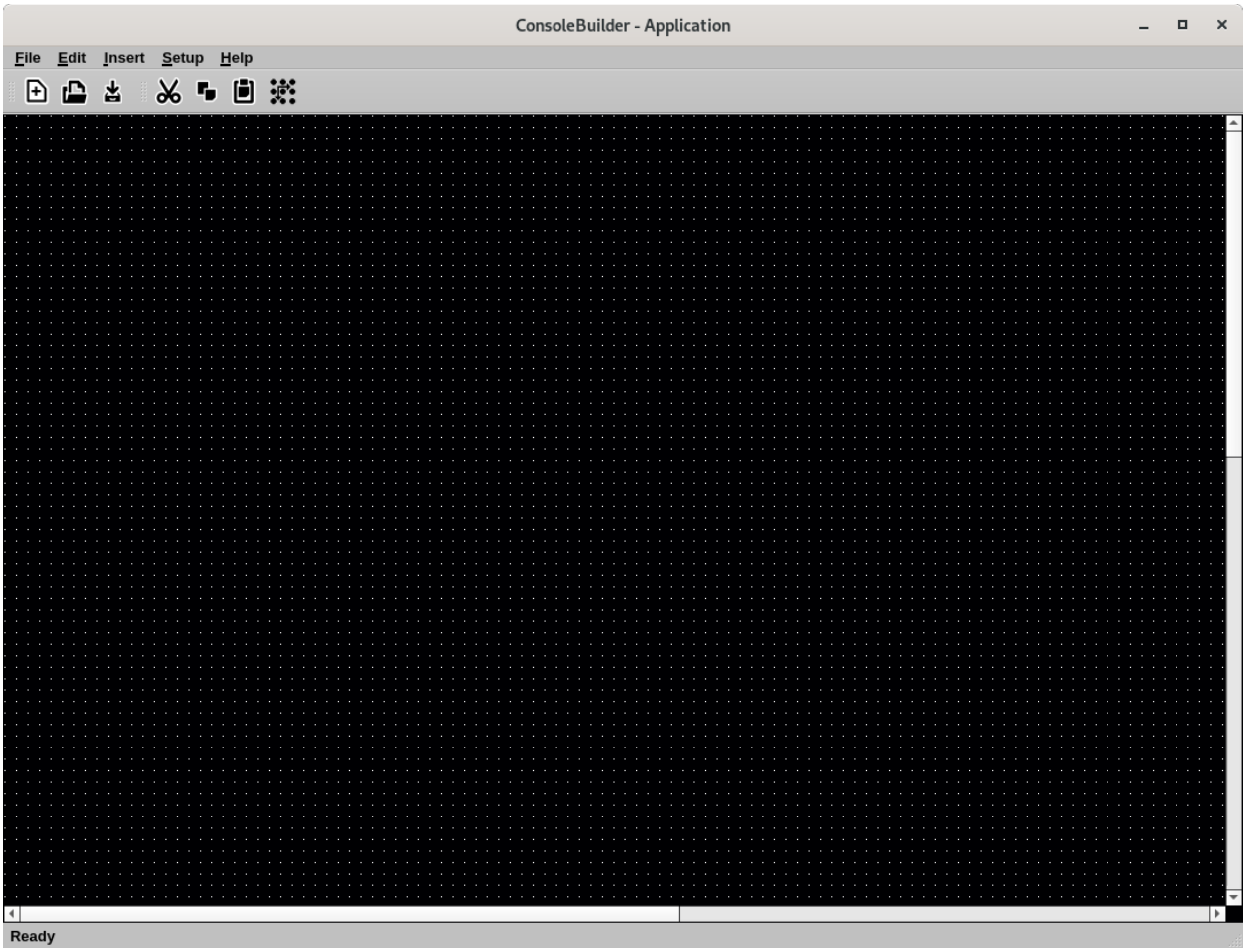


Select the **IPv4** tab. Assign an address by clicking **Manual** and input the appropriate network information. Interoperability testing used **Automatic (DHCP)**.



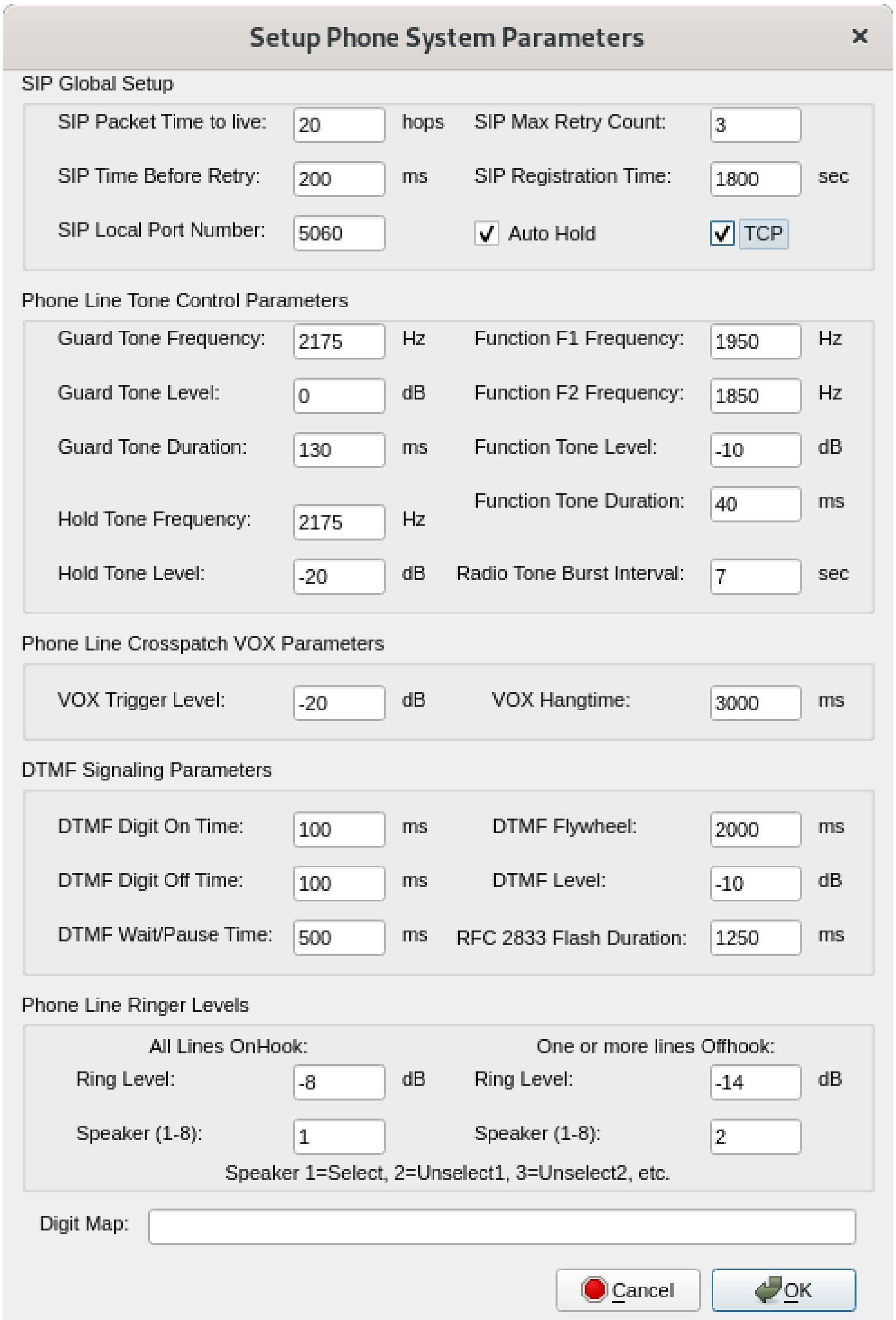
7.2. Launch Console Builder

The Console user interface is configured using Console Builder, an application resident on the MaxPlus Dispatch Console environment. Launch the application from the Administrator account on the system through the **Applications→Mindshare→ConsoleBuilder** selection.



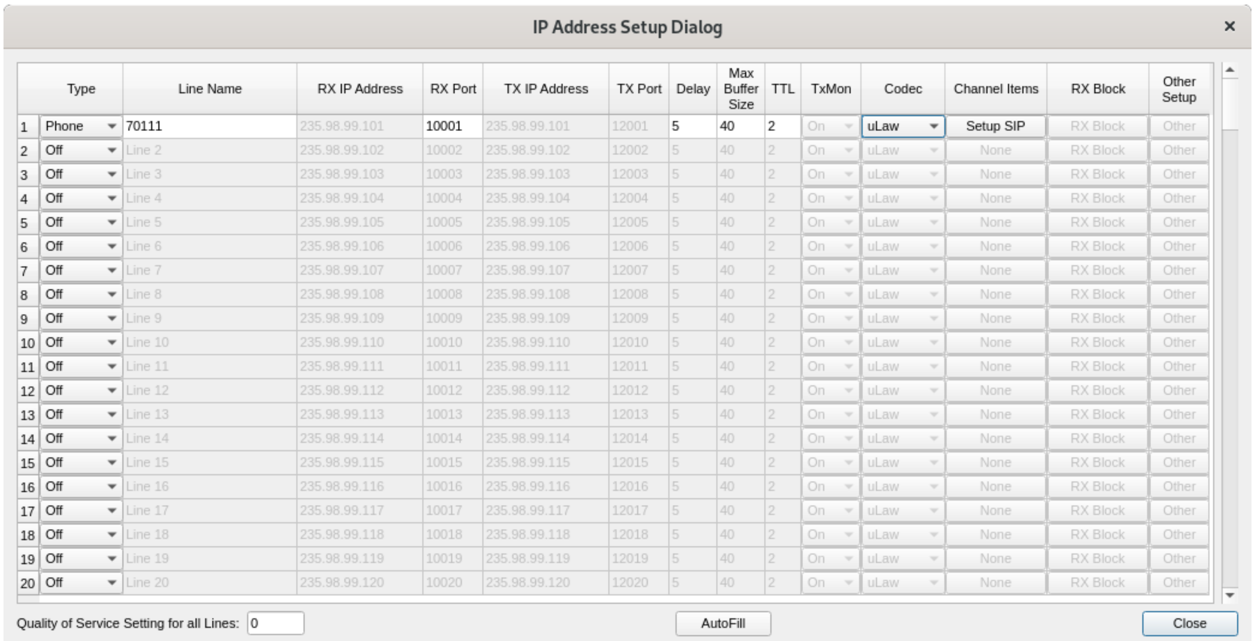
7.3. Configure Phone System Parameters

Select **Setup → Setup Phone System** from the menu. Input **20** hops for the **Set SIP Packet Time to live** entry. Select the **TCP** checkbox. The **SIP Time Before Retry** is set to **200 ms** by default but can be adjusted to 2000 ms**.** Interoperability testing used 200 ms. The **Digit Map** can specify valid extensions in the system, e.g., **7xxxxx**, if desired. Interoperability testing did not specify mapping. Retain the default values in the remaining fields.



7.4. Administer IP Comms

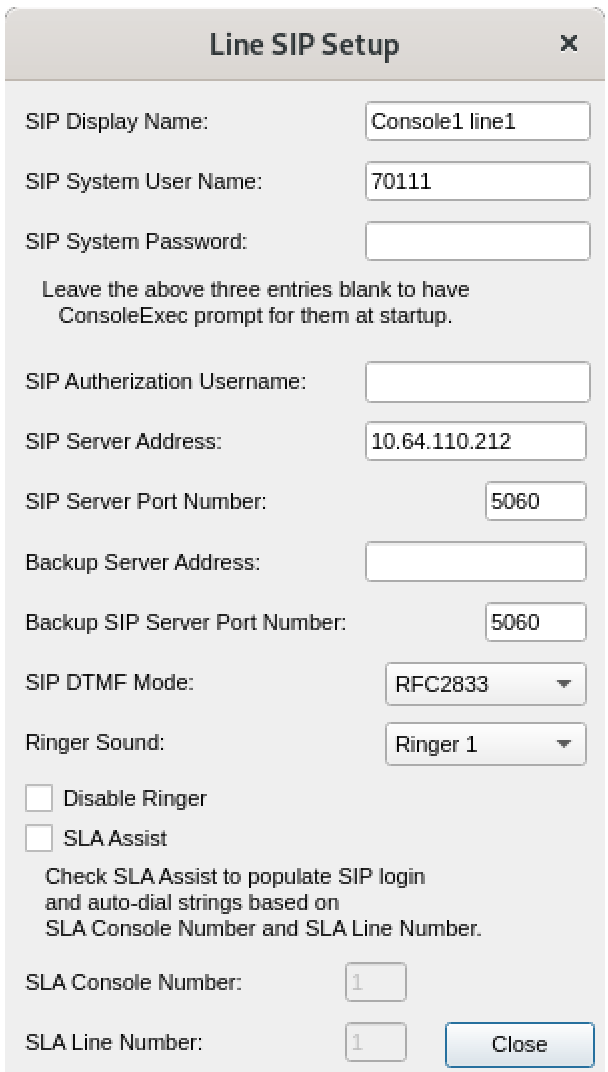
Select **Setup → Setup IP Comms** from the menu to administer lines on the console. Select **Phone** in the **Type** column. Input **70111** in the **Line Name** column. Select **uLaw** or **G.729** for the **Codec** column.



Click **Setup SIP** in the **Channel Items** column to open the **Line SIP Setup dialog**. Enter the following:

- **SIP Display Name:** Enter a name, e.g., **Console1 line1**.
- **SIP System User Name:** Enter the SIP username from **Section 6.3.2** e.g., **70111**
- **SIP System Password:** Enter the password for the user from **Section 6.3.3**.
- **SIP Server Address:** Enter the Session Manager IP address e.g., **10.64.110.212**
- **SIP Server Port Number:** Enter **5060**.
- **Backup SIP Server Port Number:** Enter **5060**.

Retain the default values in the remaining fields.

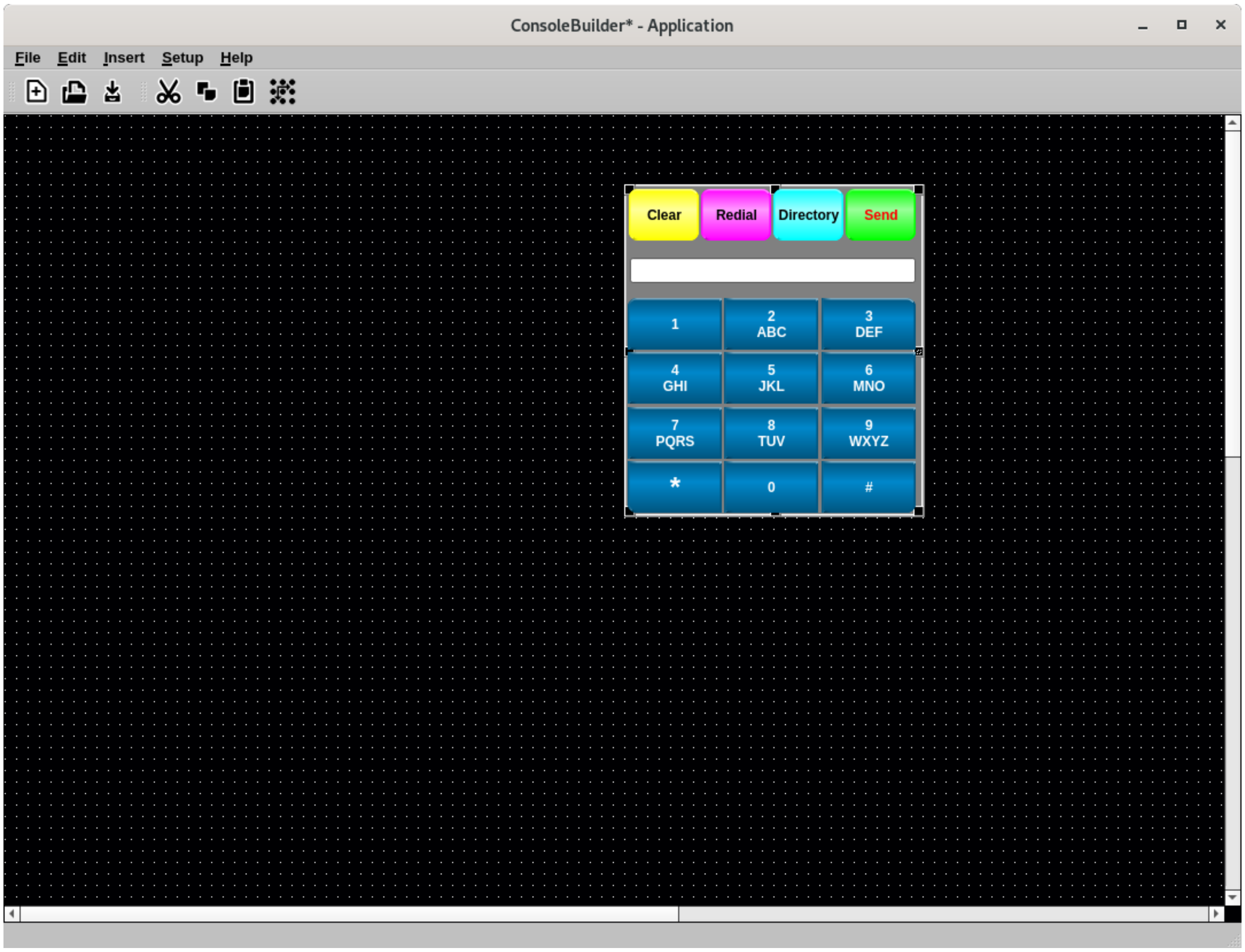


7.5. Create Console Layout

Insert and configure console elements used for the VoIP user. Elements include the Dial Keypad, Line Indicator, Phone Line, and Feature Buttons. Feature buttons employed for interoperability testing included Mute, Hold, Transfer, Blind Transfer, Call Pickup, Call Park, and Answer Back (Call Unpark) buttons. Console builder button controls provides a set of User Interface Functions that assign to buttons. These tools can be employed to provide additional functionality. A possible Call Park and Answer Back button configuration is shown below. For other button implementation configurations consult refer to **[1]** for details

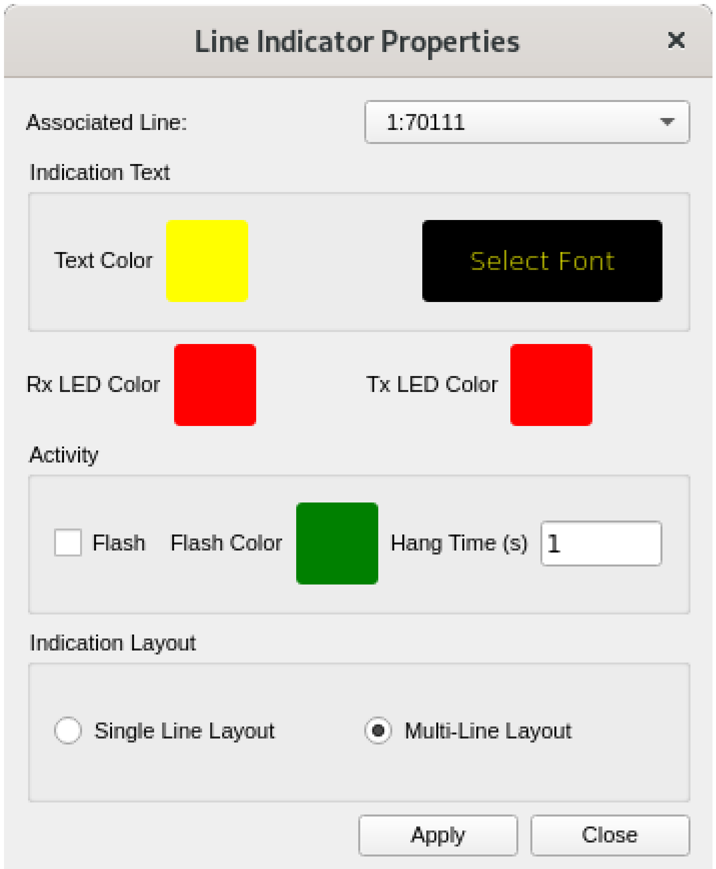
7.5.1. Dial Keypad

Select **Insert → Insert DTMF Keypad**. Adjust the size and position of the keypad on the grid.



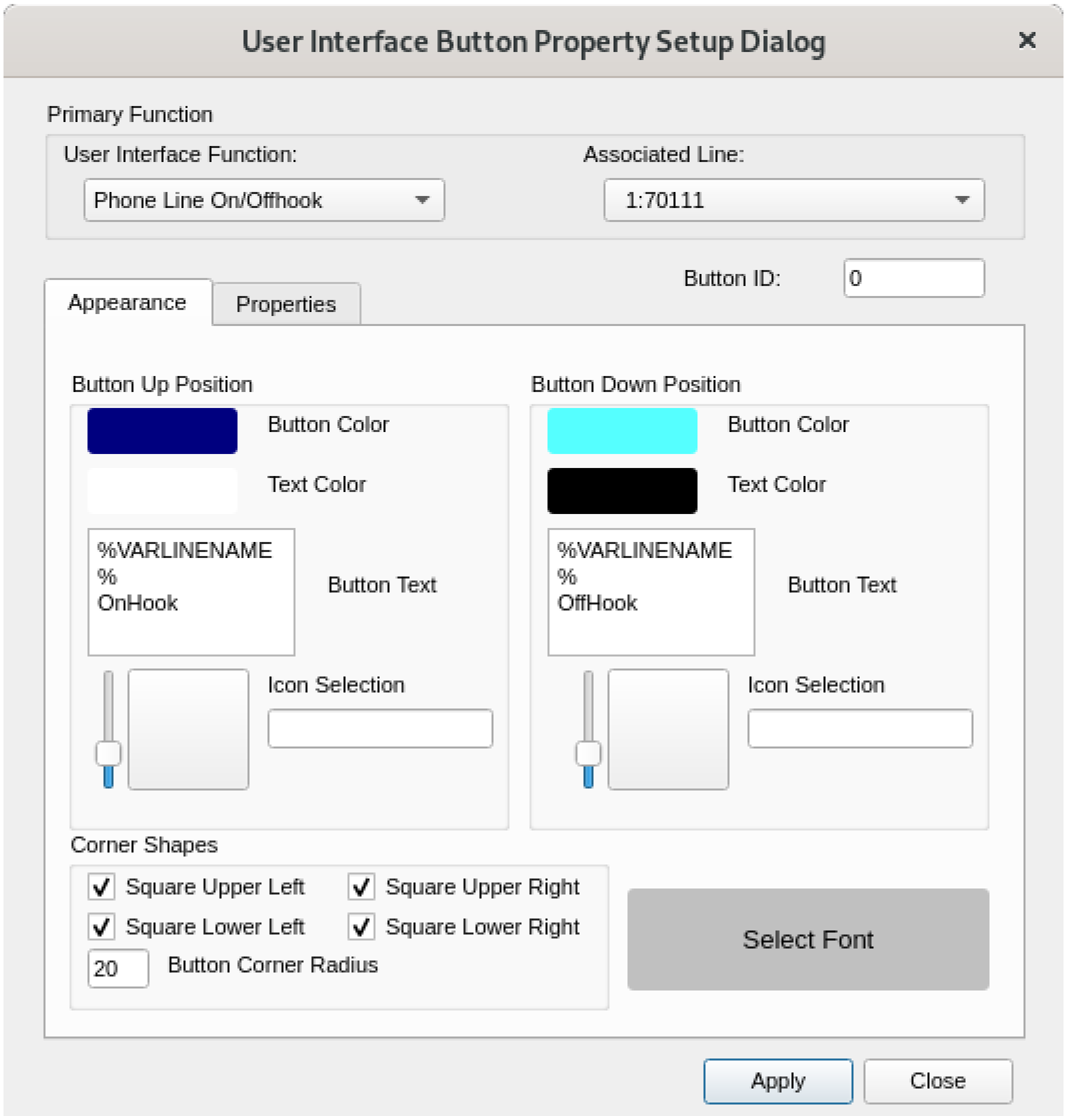
7.5.2. Line Indicator

Select **Insert → Insert Line Indicator**. Adjust the size and position of the keypad on the grid. Mouse over the Line Indicator and right click on **Properties**. Select the Line defined in **Section 7.3** for **Associated Line**, e.g., **70111**.



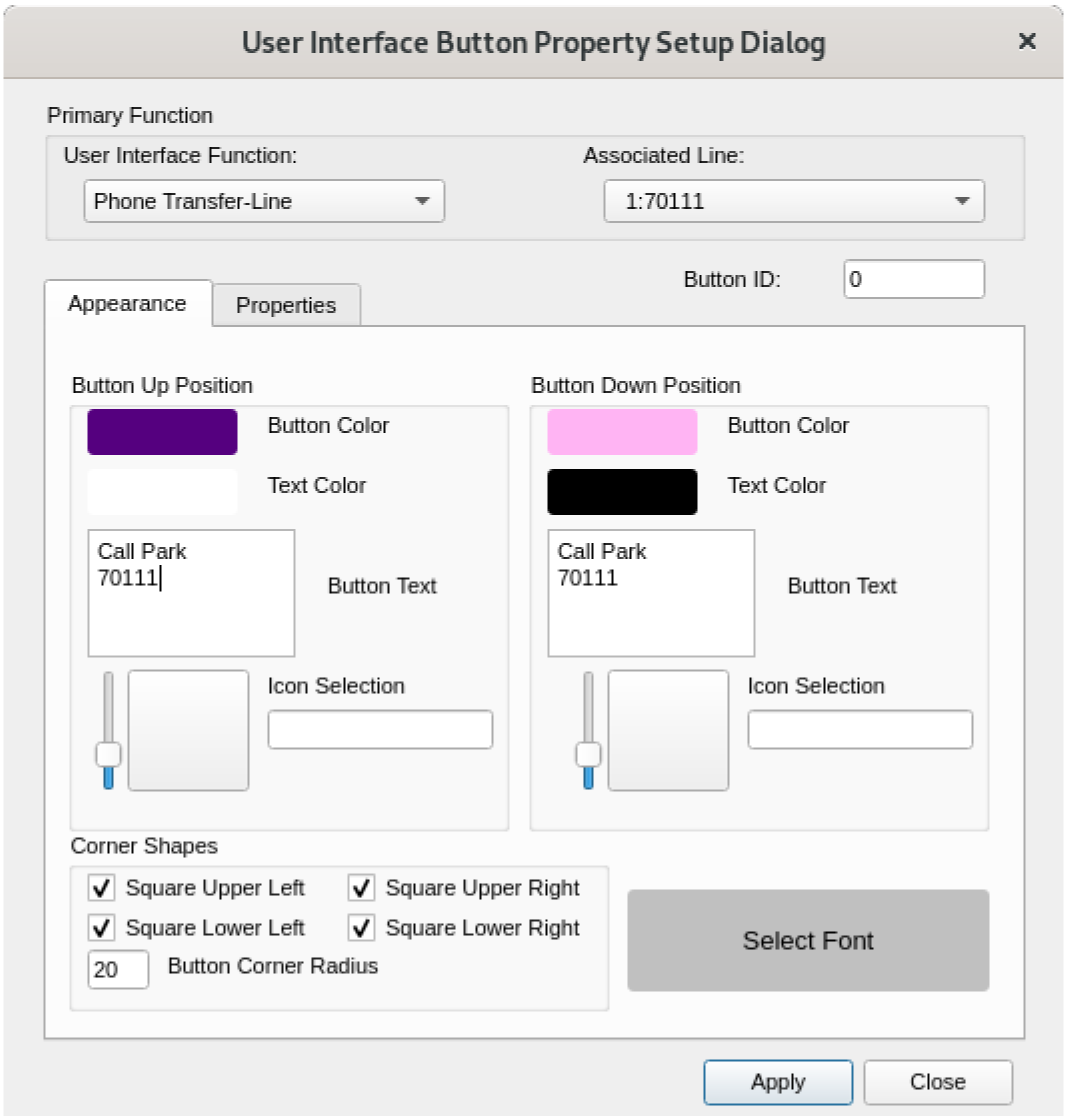
7.5.3. Phone Line On/Offhook Button

Select **Insert → Insert Button Control**. Adjust the size and position of the keypad on the grid. Mouse over the new button and right click on **Properties**. Select the Line defined in **Section 7.3** for **Associated Line**, e.g., **70111**. Select **Phone Line On/Offhook** for **User Interface Function**. Input **%VARLINENAME%[On|Off]Hook** for **Button Text** to display the Line Name of the Associated Line selected above.

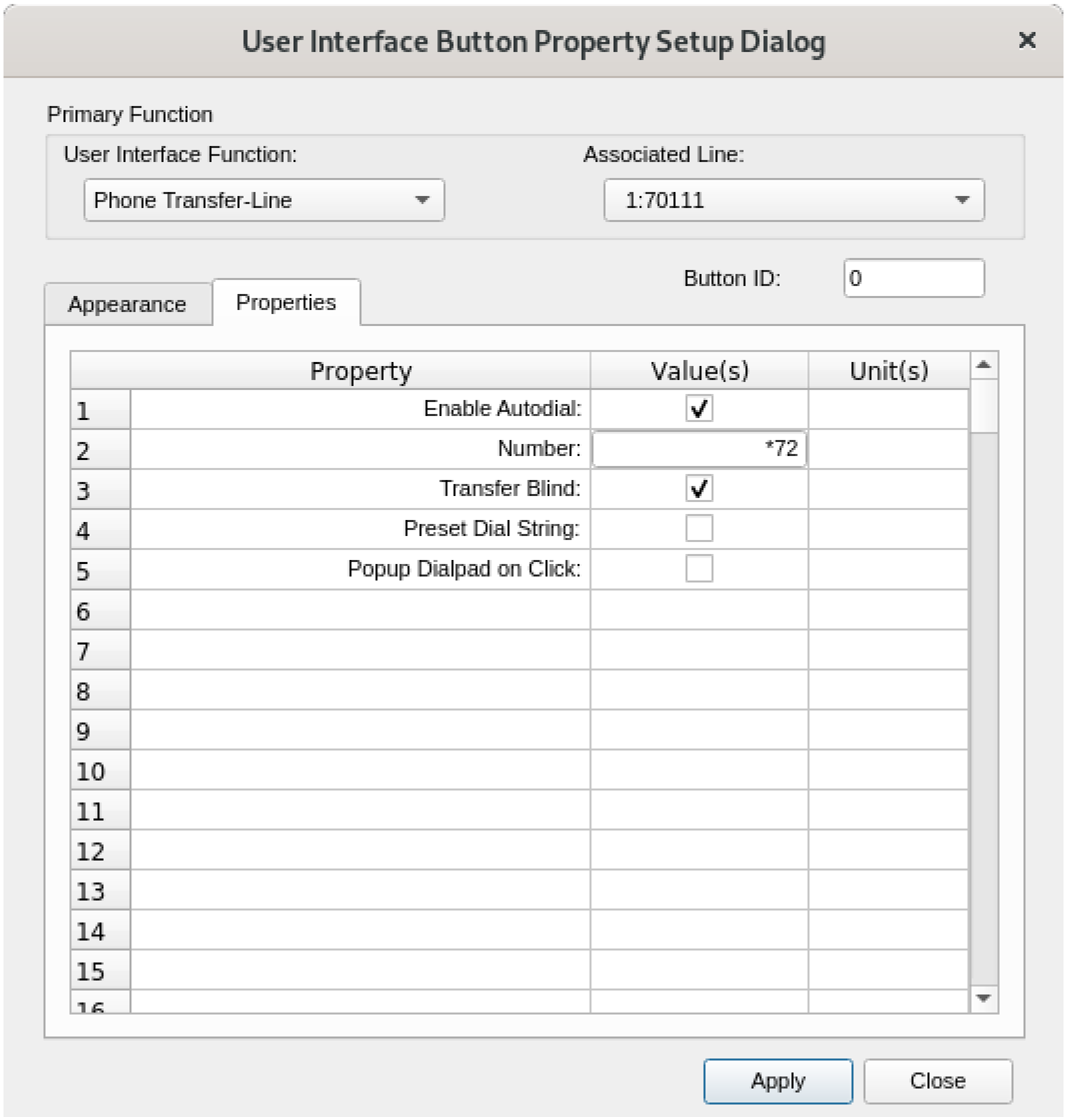


7.5.4. Call Park Button

Note: In order for this particular button implementation to work, Communication Manager Class of Service **Console Permissions** should be set to '**n**'. Calls will park to the station's extension by default. Refer to **[1]** for Console Permissions scope and details to determine applicability.
Select **Insert → Insert Button Control**. Adjust the size and position of the button control on the grid. Mouse over the new button and right click on **Properties**. Select the Line defined in **Section 7.3** for **Associated Line**, e.g., **70111**. Select **Phone Transfer-Line** for **User Interface Function**. Input appropriate **Button Text**.

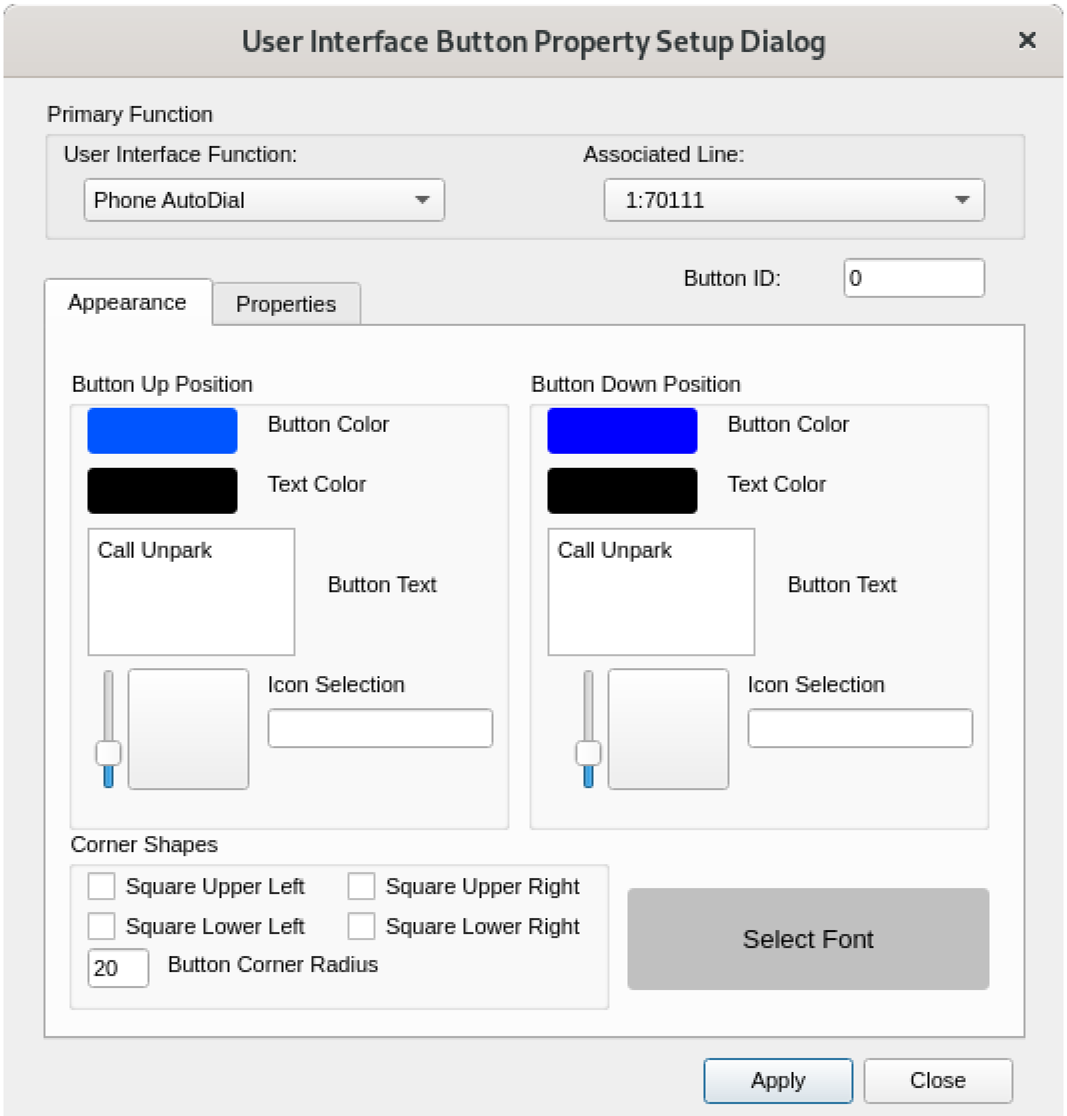


Select the **Properties** tab. Check **Enable Autodial** and **Transfer Blind**. Input the Feature Access Code assigned to Call Park on Communication Manager, e.g., ***72**.

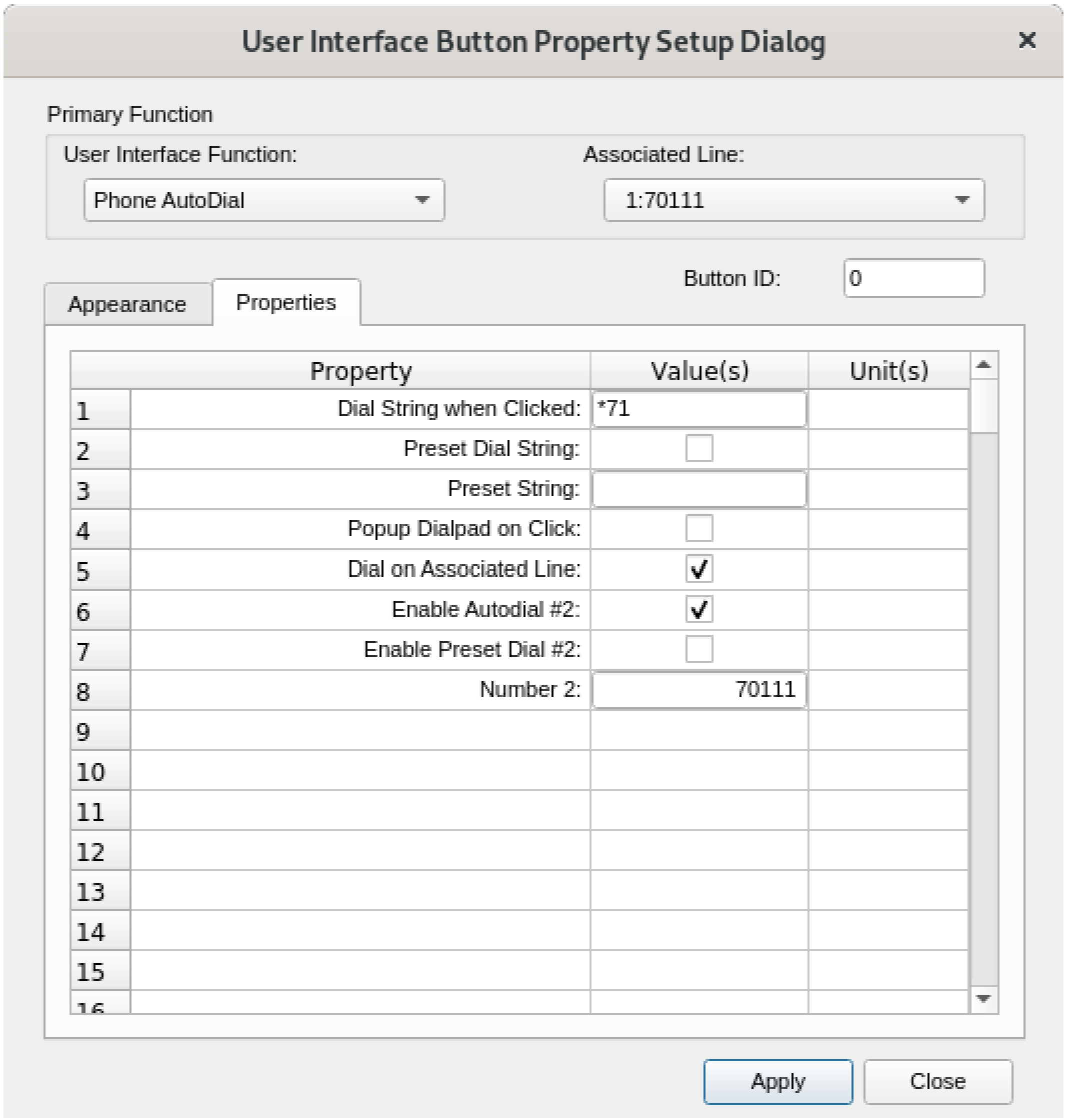


7.5.5. Answer Back (Call Unpark) Button

Select **Insert → Insert Button Control**. Adjust the size and position of the button control on the grid. Mouse over the new button and click properties. Select the Line defined in **Section 7.3** for **Associated Line**, e.g., **70111**. Select **Phone AutoDial** for **User Interface Function**. Input appropriate **Button Text**.

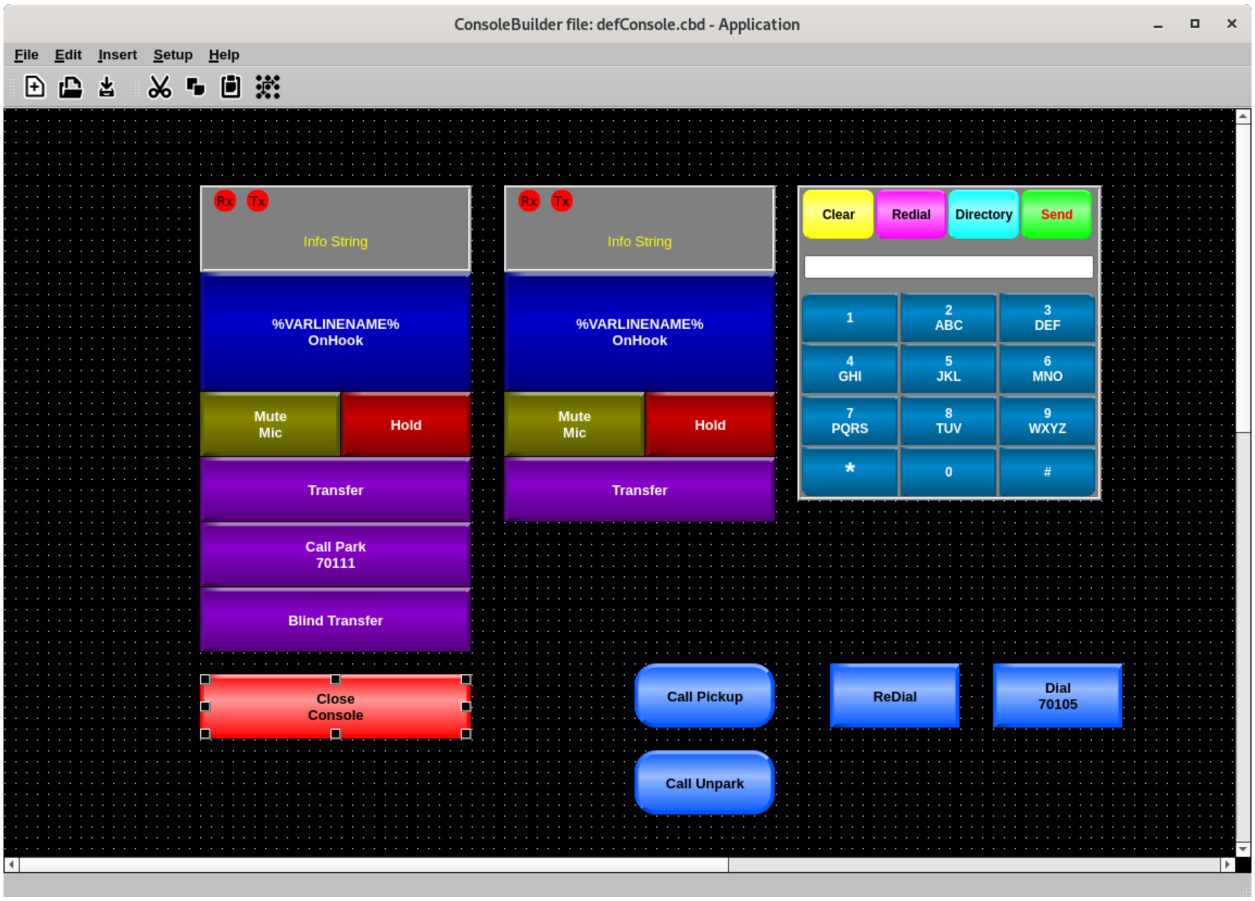


Select the **Properties** tab. Check **Dial on Associated Line** and **Enable Autodial #2**. Input the Feature Access Code assigned to Call Park on Communication Manager to **Dial String when Clicked**, e.g., ***71**. Input the line extension for **Number 2**, e.g., **70111**.



7.6. Save Layout to Configuration File

When the layout is complete, select **File → Save As** to save the layout configuration. The configuration file should be saved as **/opt/mindshare/consolesuite/defConsole.cbd**. The layout used for Interoperability tests included a second extension.

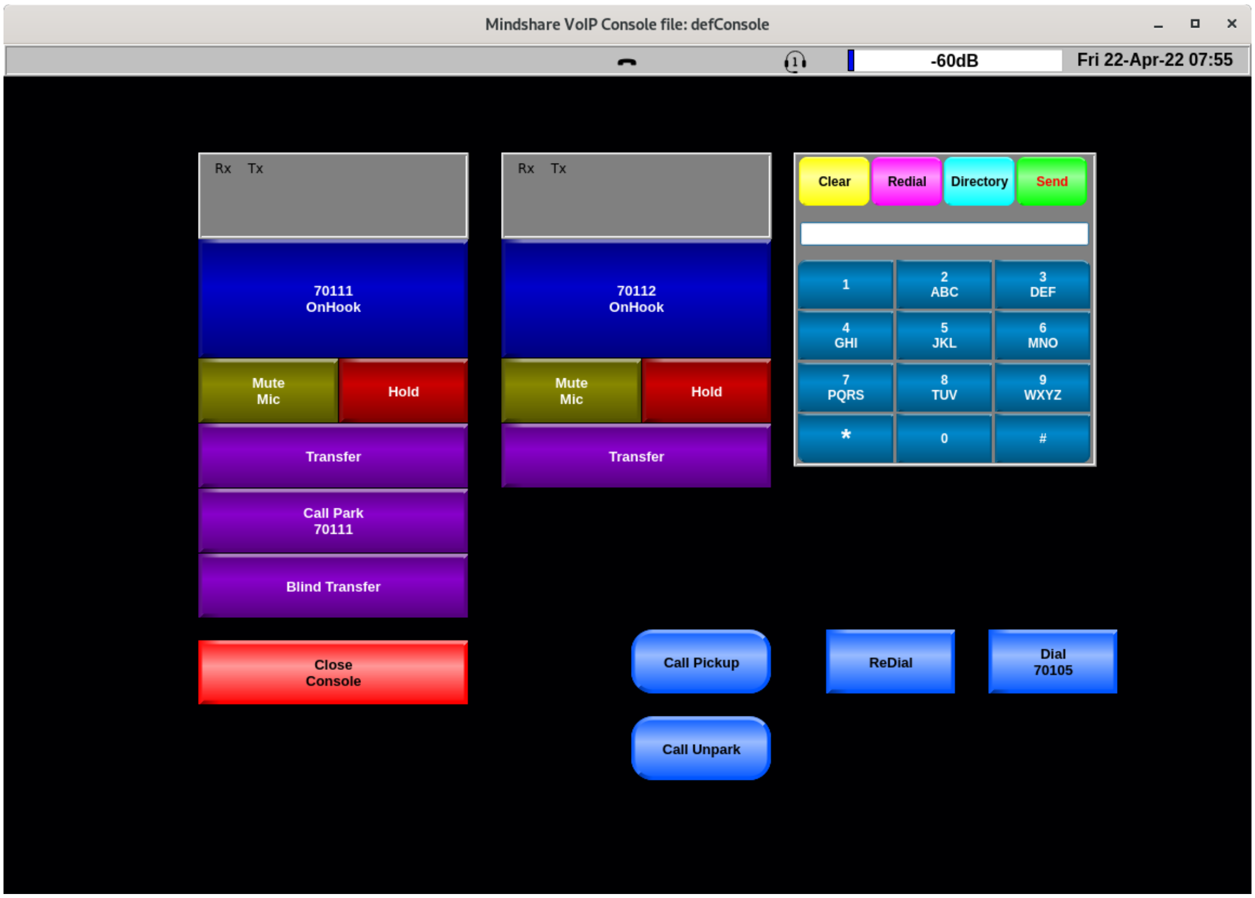


8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of MaxPlus Dispatch Console with Communication Manager and Session Manager.

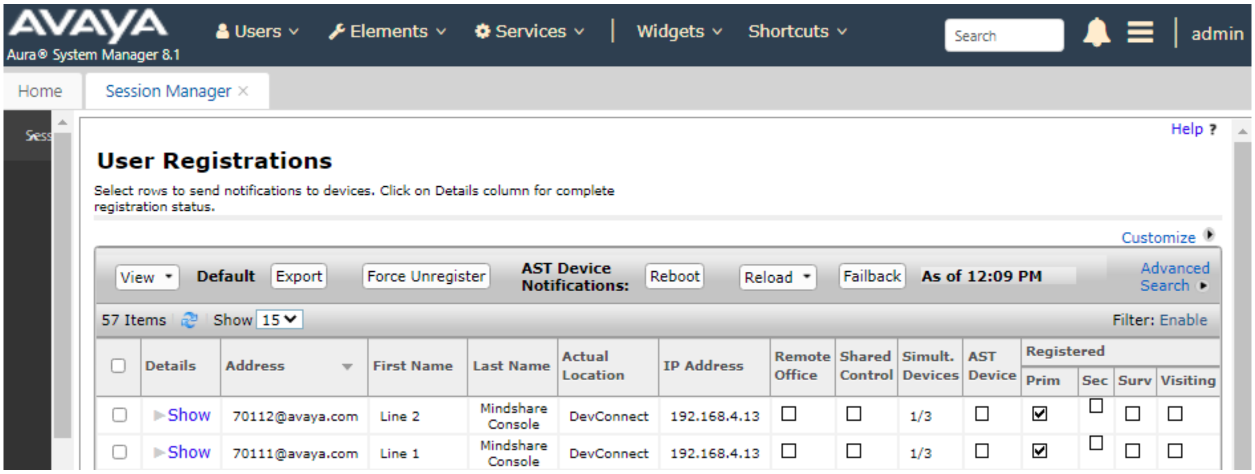
8.1. Launch Console

Launch the application from the Administrator account on the system through the **Applications→Mindshare→ConsoleBuilder** menu selection.



8.2. Registration to Session Manager

Verify that MaxPlus Dispatch Console has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status. Two lines for the console are registered as shown in the **Prim** column. The **AST Device** column is not checked.



8.3. Basic Calls

Verify basic telephony features by initiating and answering calls between MaxPlus Dispatch Console, Avaya H.323 phones, Avaya SIP phones, and other MaxPlus Dispatch Consoles.

9. Conclusion

These Application Notes have described the administration steps required to integrate MaxPlus 100500 Dispatch Console 3.27.2 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. CSS Mindshare 100500 MaxPlus Dispatch Console successfully registered with Session Manager as a SIP user and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section** Error! Reference source not found..

10. Additional References

This section references product documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Issue 12, Release 8.1.x, July 2021
[2] *Administering Avaya Aura® Session Manager*, Issue 10, Release 8.1.x, September 2021

The following CSS Mindshare product documentation is accessible to registered users at customer.css-mindshare.com.

[3] *MS0101_UM_ConsoleApplicationManual*

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.