



## Avaya Solution & Interoperability Test Lab

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# 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.

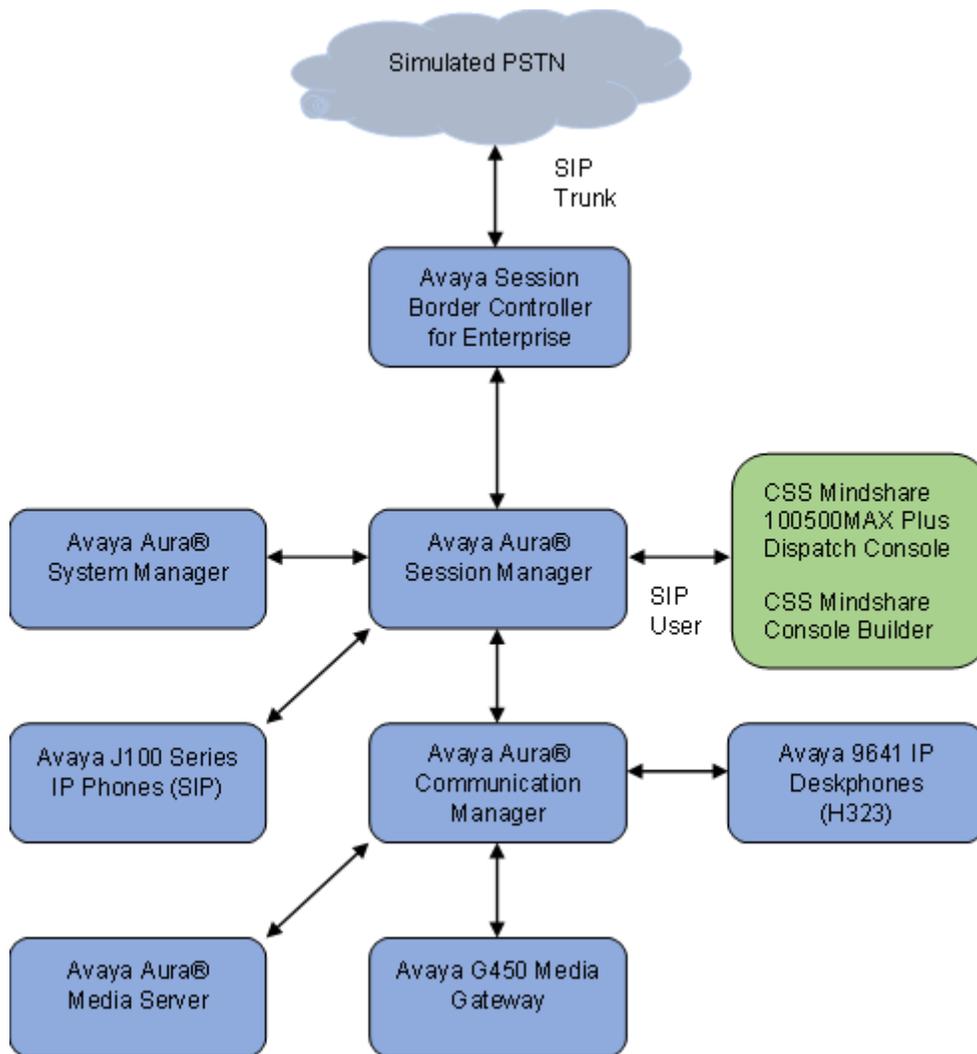
### 2.3. Support

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
H.323 IP ENDPOINTS   AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y      RSVP Enabled? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

## 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 SRTP: 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.

Recommended access to System Manager is via FQDN.  
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

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This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID:

Password:

[Change Password](#)

**Supported Browsers:** Internet Explorer 11.x or Firefox (minimum version 65.0).

## 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.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'SIP Entity Details' and contains the following fields:

- Name:** sm81
- IP Address:** 10.64.110.212
- SIP FQDN:** (empty)
- Type:** Session Manager
- Notes:** (empty)
- Location:** DevConnect
- Outbound Proxy:** (empty)
- Time Zone:** America/Denver
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)

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.

**Listen Ports**

Add Remove

4 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	<input type="text"/>
<input type="checkbox"/>	5062	TLS	avaya.com	<input type="checkbox"/>	<input type="text"/>

Select : All, None

## 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**).

The screenshot displays the 'User Profile | Add' form in the Avaya Aura System Manager 8.1 interface. The form is organized into several sections:

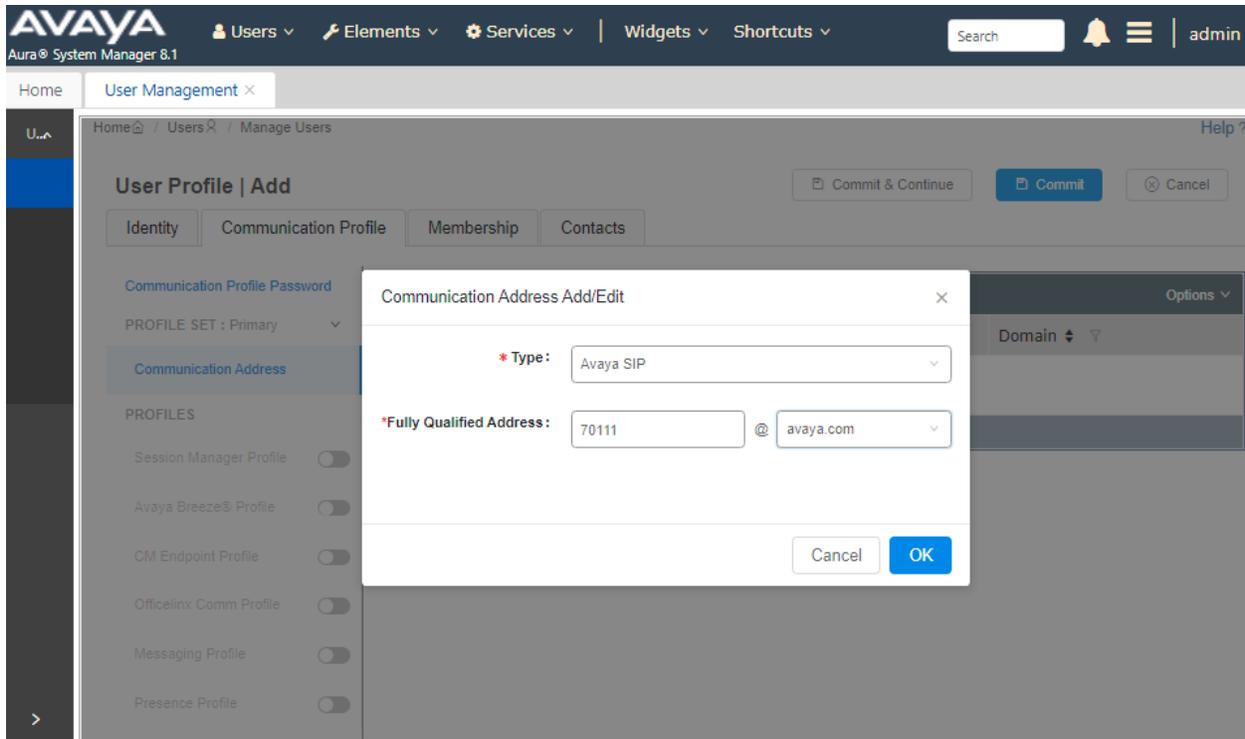
- User Provisioning Rule:** A dropdown menu.
- Identity Section:** Includes fields for:
  - \* Last Name:** Mindshare Console
  - Last Name (in Latin alphabet characters):** Mindshare Console
  - \* First Name:** Line 1
  - First Name (in Latin alphabet characters):** Line 1
  - \* Login Name:** 70111@avaya.com
  - Middle Name:** Middle Name Of User
  - Description:** Description Of User
  - Email Address:** Email Address Of User
  - Password:** (empty)
  - User Type:** Basic
  - Confirm Password:** (empty)
  - Localized Display Name:** Localized Display Name Of

### 6.3.2. Communication Address

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

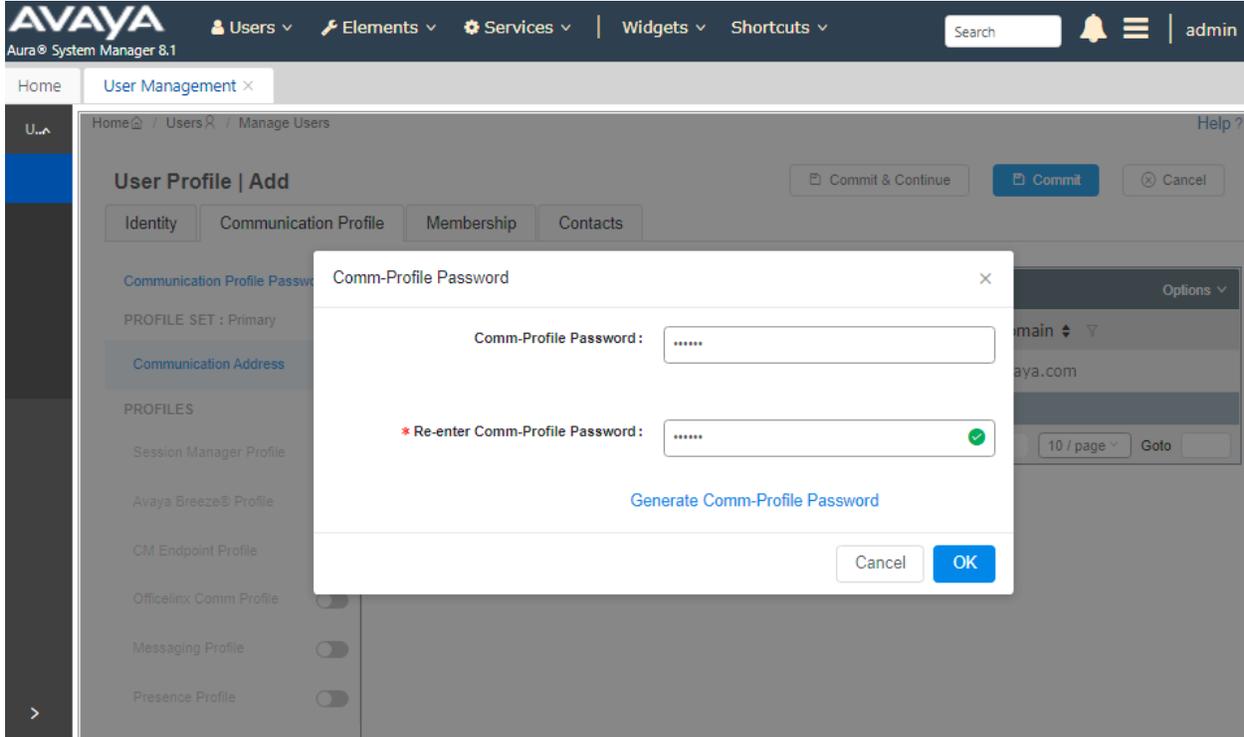
Enter the following attributes for the **Communication Address**:

- **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**.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, a search bar, and a user profile 'admin'. The main content area is titled 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Session Manager Profile' toggle switch turned on. Below this, the 'SIP Registration' section contains the following fields: 'Primary Session Manager' (value: sm81), 'Secondary Session Manager' (value: Start typing...), 'Survivability Server' (value: Start typing...), 'Max. Simultaneous Devices' (value: 3), and 'Block New Registration When Maximum Registrations Active?' (checked). The 'Application Sequences' section contains 'Origination Sequence' (value: cm81) and 'Termination Sequence' (value: cm81). Action buttons 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right of the form.

### 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).

The screenshot displays the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts' menus, along with a search bar and a user profile icon labeled 'admin'. The main content area is titled 'User Profile | Add' and features a sidebar with profile categories: 'Communication Profile Password', 'PROFILES', and 'Session Manager Profile'. The 'CM Endpoint Profile' is selected and active. The main form area contains the following fields and controls:

- System:** A dropdown menu with 'cm81' selected.
- Profile Type:** A dropdown menu with 'Endpoint' selected.
- Extension:** A text input field containing '70111' with an 'Endpoint Editor' icon to its right.
- Template:** A dropdown menu with 'J179\_DEFAULT\_CM\_8\_1' selected.
- Set Type:** A text input field containing 'J179'.
- Port:** A dropdown menu with 'IP' selected.
- Security Code:** A text input field with the placeholder 'Enter Security Code'.
- Preferred Handle:** A dropdown menu with 'Select' selected.
- Voice Mail Number:** An empty text input field.
- Sip Trunk:** An empty text input field.
- Use Existing Endpoints:** An unchecked checkbox.
- Calculate Route Pattern:** An unchecked checkbox.
- SIP URI:** A dropdown menu with 'Select' selected.
- Delete on Unassign from User or on Delete User:** A checked checkbox.
- Override Endpoint Name and Localized Name:** A checked checkbox.
- Allow H.323 and SIP Endpoint Dual Registration:** An unchecked checkbox.

At the top right of the form area, there are three buttons: 'Commit & Continue', 'Commit', and 'Cancel'.

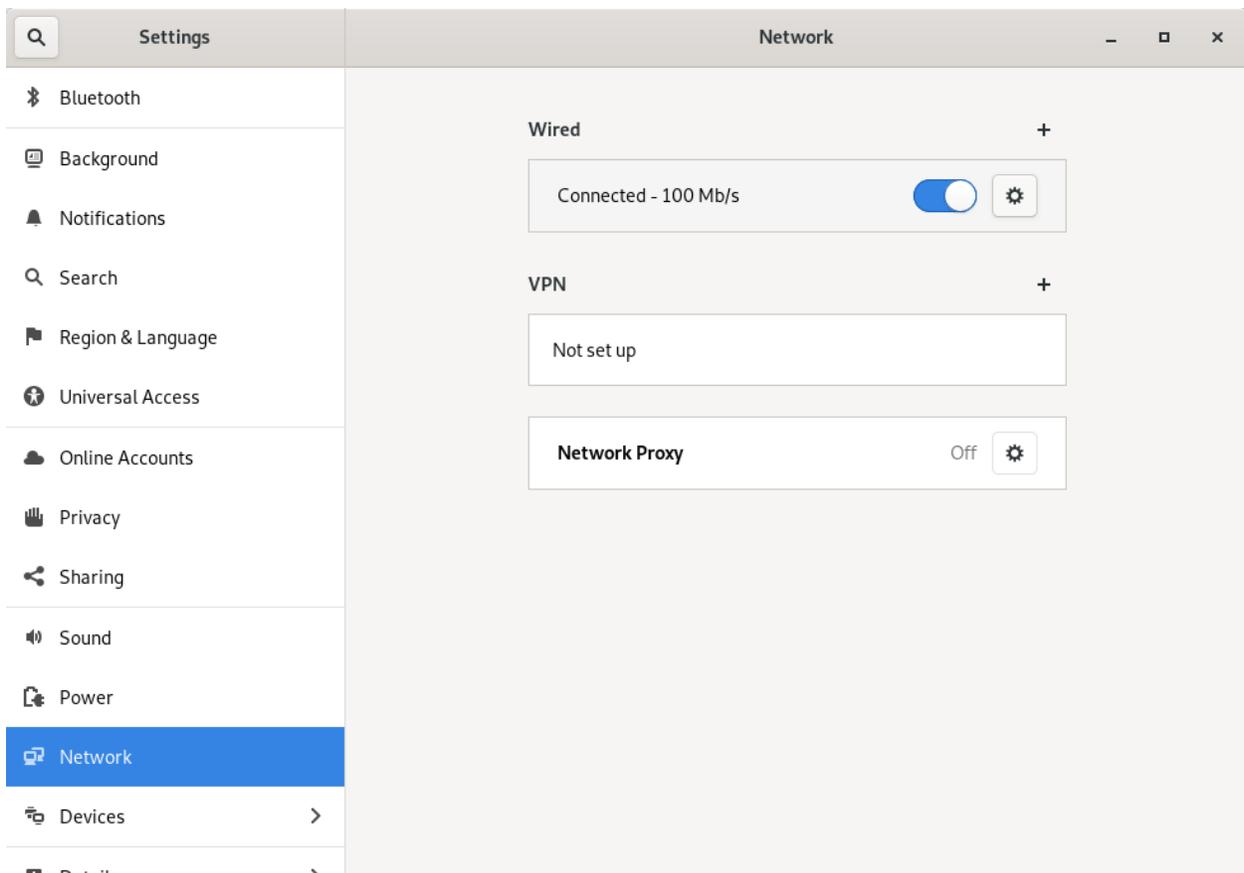
## 7. Configure CSS Mindshare 100500 MaxPlus Dispatch Console

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)**.

Cancel Wired Apply

Details Identity **IPv4** IPv6 Security

**IPv4 Method**

Automatic (DHCP)  Link-Local Only  
 Manual  Disable

**DNS** Automatic

Separate IP addresses with commas

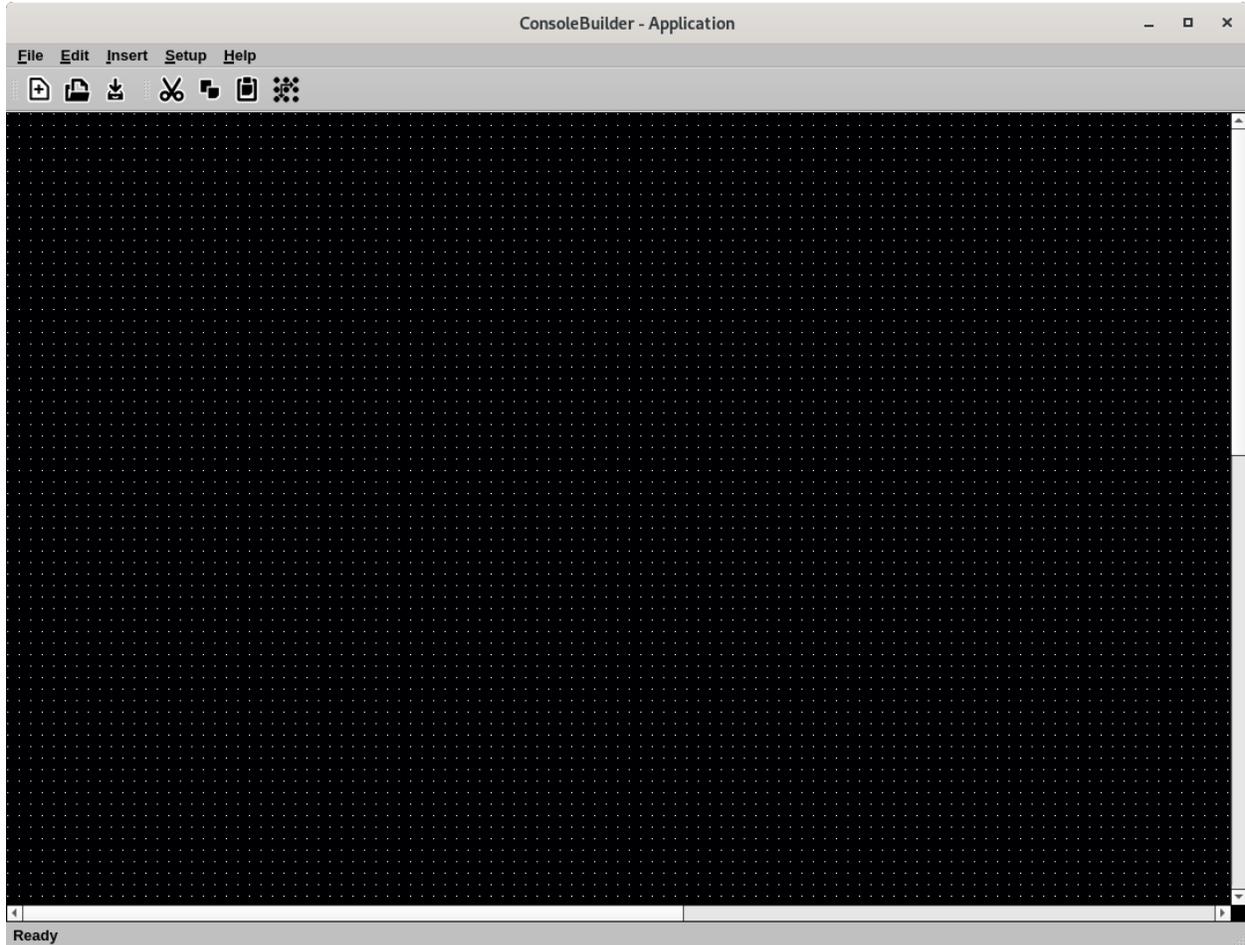
**Routes** Automatic

Address	Netmask	Gateway	Metric	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="x"/>

Use this connection only for resources on its network

## 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.

**Setup Phone System Parameters** ✕

**SIP Global Setup**

SIP Packet Time to live: <input type="text" value="20"/> hops	SIP Max Retry Count: <input type="text" value="3"/>
SIP Time Before Retry: <input type="text" value="200"/> ms	SIP Registration Time: <input type="text" value="1800"/> sec
SIP Local Port Number: <input type="text" value="5060"/>	<input checked="" type="checkbox"/> Auto Hold <input checked="" type="checkbox"/> TCP

**Phone Line Tone Control Parameters**

Guard Tone Frequency: <input type="text" value="2175"/> Hz	Function F1 Frequency: <input type="text" value="1950"/> Hz
Guard Tone Level: <input type="text" value="0"/> dB	Function F2 Frequency: <input type="text" value="1850"/> Hz
Guard Tone Duration: <input type="text" value="130"/> ms	Function Tone Level: <input type="text" value="-10"/> dB
Hold Tone Frequency: <input type="text" value="2175"/> Hz	Function Tone Duration: <input type="text" value="40"/> ms
Hold Tone Level: <input type="text" value="-20"/> dB	Radio Tone Burst Interval: <input type="text" value="7"/> sec

**Phone Line Crosspatch VOX Parameters**

VOX Trigger Level: <input type="text" value="-20"/> dB	VOX Hangtime: <input type="text" value="3000"/> ms
--	--

**DTMF Signaling Parameters**

DTMF Digit On Time: <input type="text" value="100"/> ms	DTMF Flywheel: <input type="text" value="2000"/> ms
DTMF Digit Off Time: <input type="text" value="100"/> ms	DTMF Level: <input type="text" value="-10"/> dB
DTMF Wait/Pause Time: <input type="text" value="500"/> ms	RFC 2833 Flash Duration: <input type="text" value="1250"/> ms

**Phone Line Ringer Levels**

All Lines OnHook:	One or more lines Offhook:
Ring Level: <input type="text" value="-8"/> dB	Ring Level: <input type="text" value="-14"/> dB
Speaker (1-8): <input type="text" value="1"/>	Speaker (1-8): <input type="text" value="2"/>

Speaker 1=Select, 2=Unselect1, 3=Unselect2, etc.

**Digit Map:**

## 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.

**IP Address Setup Dialog** ✕

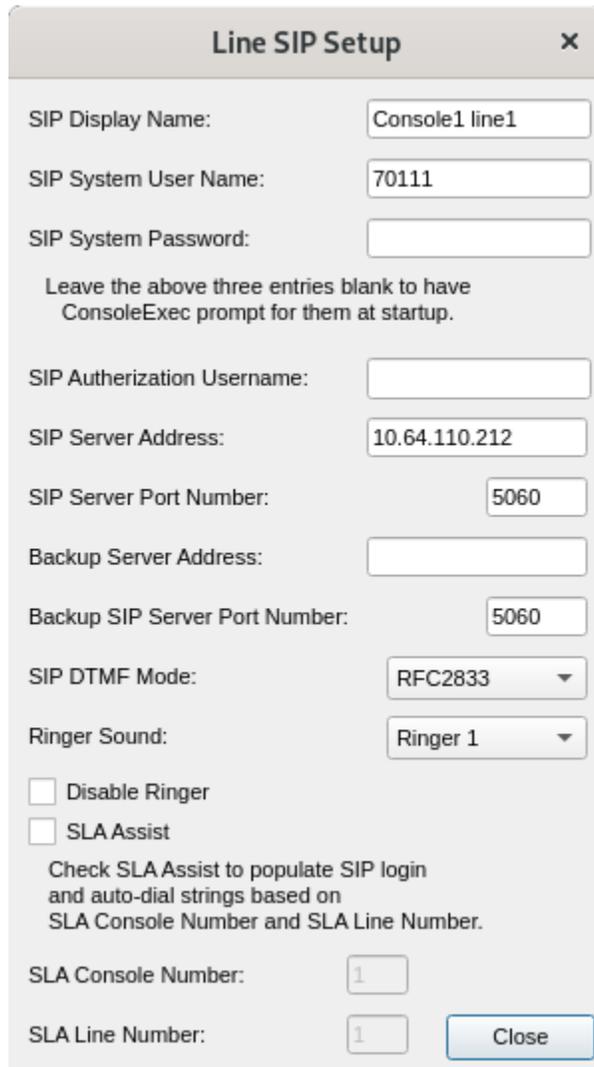
	Type	Line Name	RX IP Address	RX Port	TX IP Address	TX Port	Delay	Max Buffer Size	TTL	TxMon	Codec	Channel Items	RX Block	Other Setup
1	Phone ▾	70111	235.98.99.101	10001	235.98.99.101	12001	5	40	2	On ▾	uLaw ▾	Setup SIP	RX Block	Other
2	Off ▾	Line 2	235.98.99.102	10002	235.98.99.102	12002	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
3	Off ▾	Line 3	235.98.99.103	10003	235.98.99.103	12003	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
4	Off ▾	Line 4	235.98.99.104	10004	235.98.99.104	12004	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
5	Off ▾	Line 5	235.98.99.105	10005	235.98.99.105	12005	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
6	Off ▾	Line 6	235.98.99.106	10006	235.98.99.106	12006	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
7	Off ▾	Line 7	235.98.99.107	10007	235.98.99.107	12007	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
8	Off ▾	Line 8	235.98.99.108	10008	235.98.99.108	12008	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
9	Off ▾	Line 9	235.98.99.109	10009	235.98.99.109	12009	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
10	Off ▾	Line 10	235.98.99.110	10010	235.98.99.110	12010	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
11	Off ▾	Line 11	235.98.99.111	10011	235.98.99.111	12011	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
12	Off ▾	Line 12	235.98.99.112	10012	235.98.99.112	12012	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
13	Off ▾	Line 13	235.98.99.113	10013	235.98.99.113	12013	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
14	Off ▾	Line 14	235.98.99.114	10014	235.98.99.114	12014	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
15	Off ▾	Line 15	235.98.99.115	10015	235.98.99.115	12015	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
16	Off ▾	Line 16	235.98.99.116	10016	235.98.99.116	12016	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
17	Off ▾	Line 17	235.98.99.117	10017	235.98.99.117	12017	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
18	Off ▾	Line 18	235.98.99.118	10018	235.98.99.118	12018	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
19	Off ▾	Line 19	235.98.99.119	10019	235.98.99.119	12019	5	40	2	On ▾	uLaw ▾	None	RX Block	Other
20	Off ▾	Line 20	235.98.99.120	10020	235.98.99.120	12020	5	40	2	On ▾	uLaw ▾	None	RX Block	Other

Quality of Service Setting for all Lines:  AutoFill Close

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.



The image shows a 'Line SIP Setup' dialog box with the following fields and values:

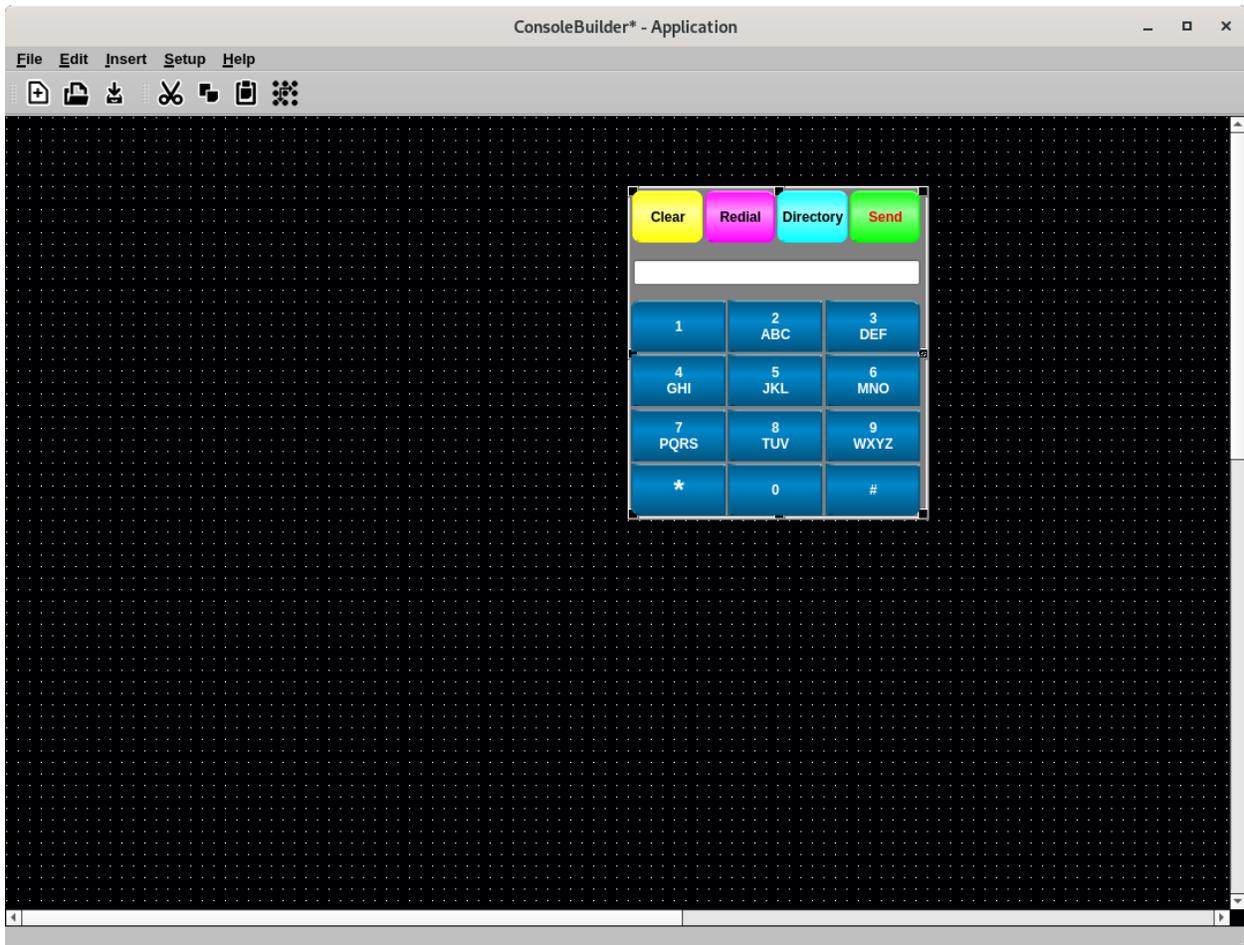
Field	Value
SIP Display Name:	Console1 line1
SIP System User Name:	70111
SIP System Password:	
Leave the above three entries blank to have ConsoleExec prompt for them at startup.	
SIP Authorization Username:	
SIP Server Address:	10.64.110.212
SIP Server Port Number:	5060
Backup Server Address:	
Backup SIP Server Port Number:	5060
SIP DTMF Mode:	RFC2833
Ringer Sound:	Ringer 1
<input type="checkbox"/> Disable Ringer	
<input type="checkbox"/> SLA Assist	
Check SLA Assist to populate SIP login and auto-dial strings based on SLA Console Number and SLA Line Number.	
SLA Console Number:	1
SLA Line Number:	1
<input type="button" value="Close"/>	

## 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

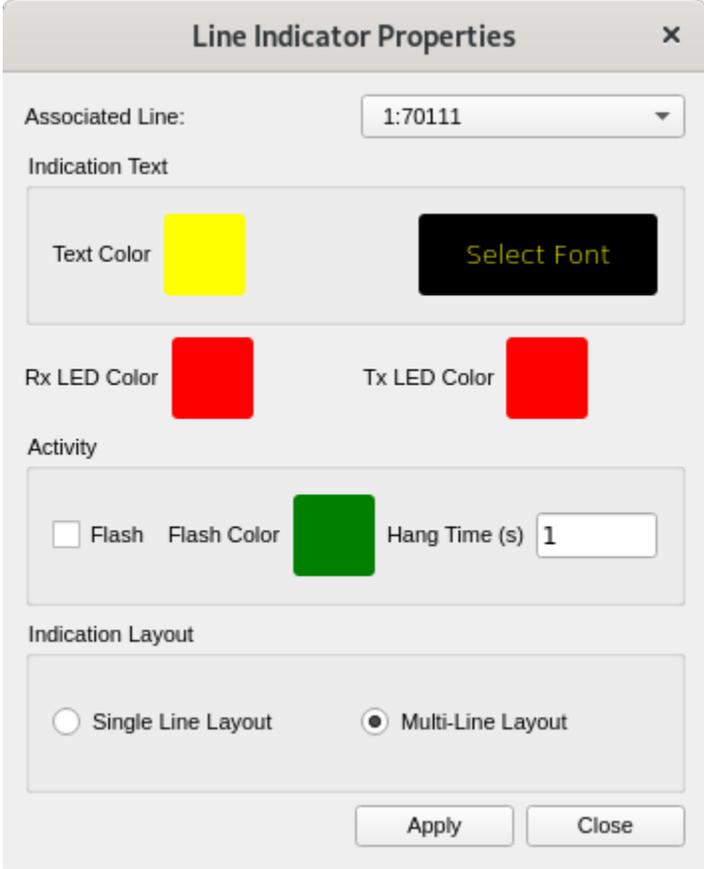
### 7.5.1. Dial Keypad

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**.



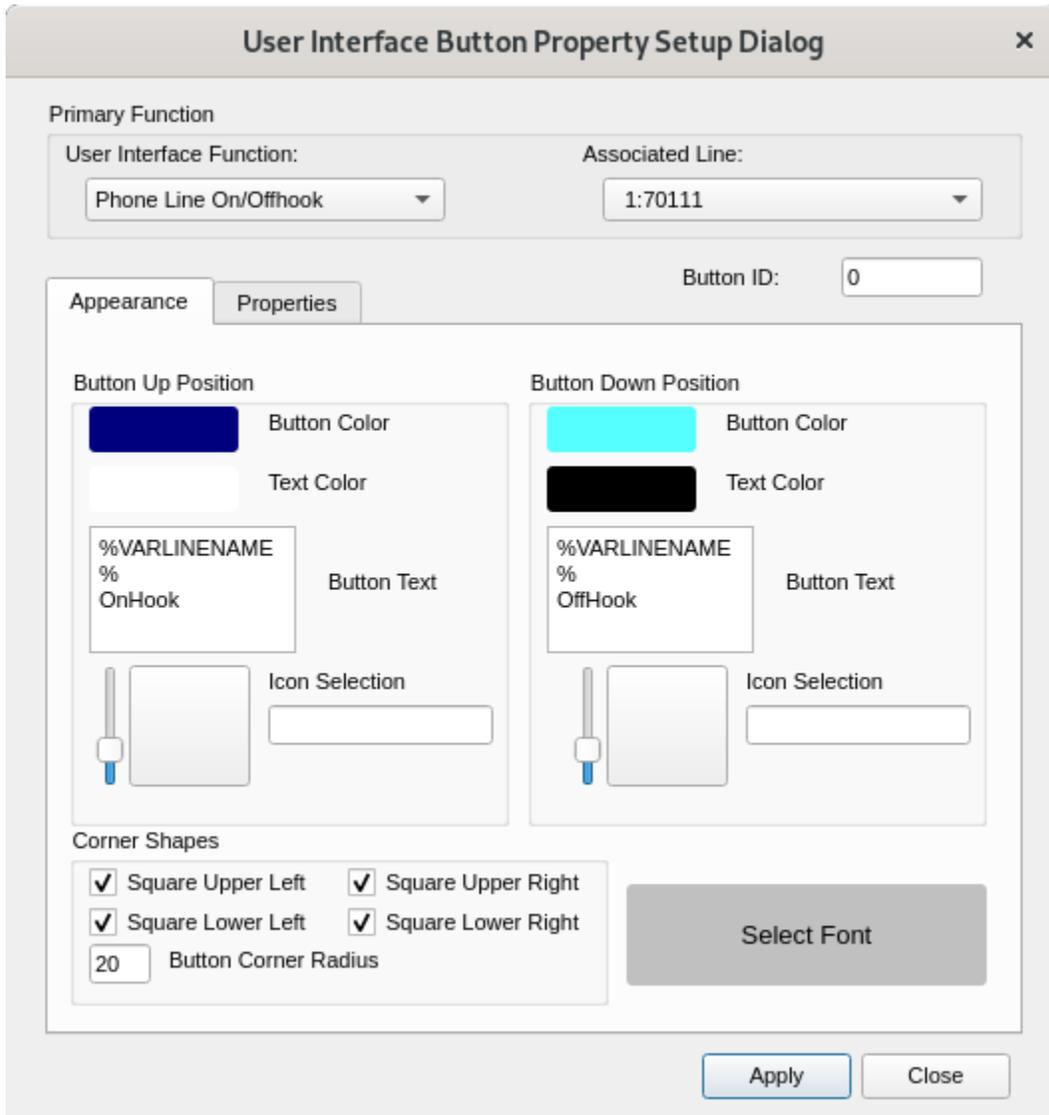
The image shows a dialog box titled "Line Indicator Properties" with a close button (X) in the top right corner. The dialog is organized into several sections:

- Associated Line:** A dropdown menu showing "1:70111".
- Indication Text:** A section containing a "Text Color" selector with a yellow square, a "Select Font" button with a black background and yellow text, an "Rx LED Color" selector with a red square, and a "Tx LED Color" selector with a red square.
- Activity:** A section containing a "Flash" checkbox (unchecked), a "Flash Color" selector with a green square, and a "Hang Time (s)" input field with the value "1".
- Indication Layout:** A section containing two radio buttons: "Single Line Layout" (unchecked) and "Multi-Line Layout" (checked).

At the bottom of the dialog are two buttons: "Apply" and "Close".

### 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.

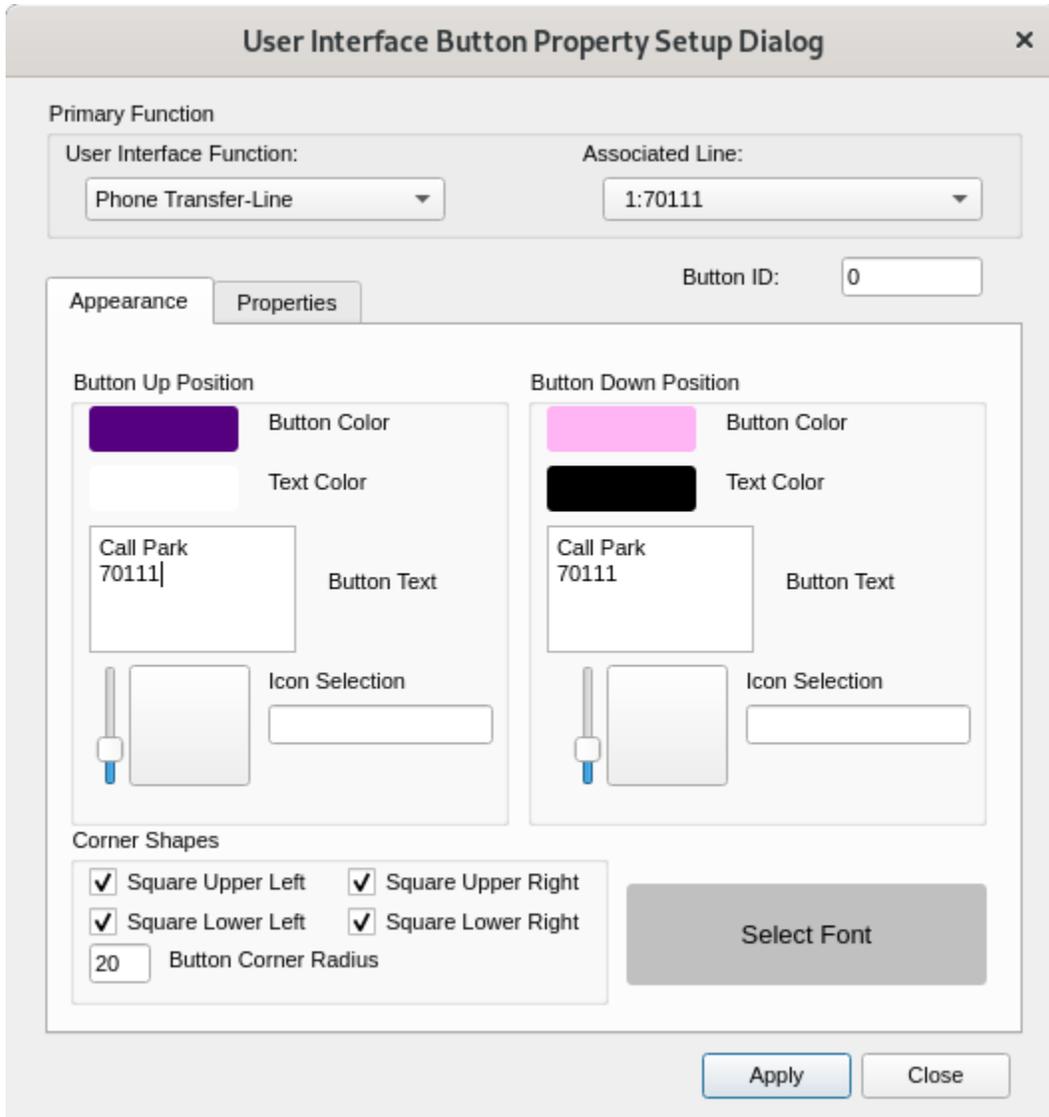


The image shows a 'User Interface Button Property Setup Dialog' window. At the top, the title bar reads 'User Interface Button Property Setup Dialog' with a close button (X) on the right. Below the title bar, there are two dropdown menus: 'User Interface Function' set to 'Phone Line On/Offhook' and 'Associated Line' set to '1:70111'. To the right of these is a 'Button ID' field containing the number '0'. Below this, there are two tabs: 'Appearance' and 'Properties', with 'Properties' currently selected. The 'Properties' tab is divided into two main sections: 'Button Up Position' and 'Button Down Position'. Each section contains a 'Button Color' swatch (dark blue for up, cyan for down), a 'Text Color' swatch (white for up, black for down), and a 'Button Text' input field containing '%VARLINENAME%' followed by '% OnHook' for the up position and '% OffHook' for the down position. Below the text fields are 'Icon Selection' controls, each consisting of a vertical slider and a square icon placeholder. At the bottom of the dialog, there is a 'Corner Shapes' section with four checked checkboxes: 'Square Upper Left', 'Square Upper Right', 'Square Lower Left', and 'Square Lower Right'. A 'Button Corner Radius' field is set to '20'. To the right of the corner shapes is a 'Select Font' button. At the very bottom of the dialog are 'Apply' and 'Close' buttons.

### 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**.



The image shows a 'User Interface Button Property Setup Dialog' window. It has a title bar with a close button (X). The dialog is divided into several sections:

- Primary Function:** Contains two dropdown menus. 'User Interface Function' is set to 'Phone Transfer-Line' and 'Associated Line' is set to '1:70111'.
- Appearance / Properties:** A tabbed interface. The 'Properties' tab is active. It includes a 'Button ID' field with the value '0'.
- Button Up Position:** A section for the button's state when not pressed. It includes:
  - 'Button Color': A purple color swatch.
  - 'Text Color': A white color swatch.
  - 'Button Text': A text input field containing 'Call Park 70111'.
  - 'Icon Selection': A vertical slider and a button icon.
- Button Down Position:** A section for the button's state when pressed. It includes:
  - 'Button Color': A pink color swatch.
  - 'Text Color': A black color swatch.
  - 'Button Text': A text input field containing 'Call Park 70111'.
  - 'Icon Selection': A vertical slider and a button icon.
- Corner Shapes:** A section with four checked checkboxes: 'Square Upper Left', 'Square Upper Right', 'Square Lower Left', and 'Square Lower Right'. Below them is a 'Button Corner Radius' field with the value '20'.
- Select Font:** A grey button labeled 'Select Font'.
- Buttons:** 'Apply' and 'Close' buttons at the bottom right.

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**.

**User Interface Button Property Setup Dialog** ✕

Primary Function

User Interface Function: Phone Transfer-Line ▾      Associated Line: 1:70111 ▾

Button ID: 0

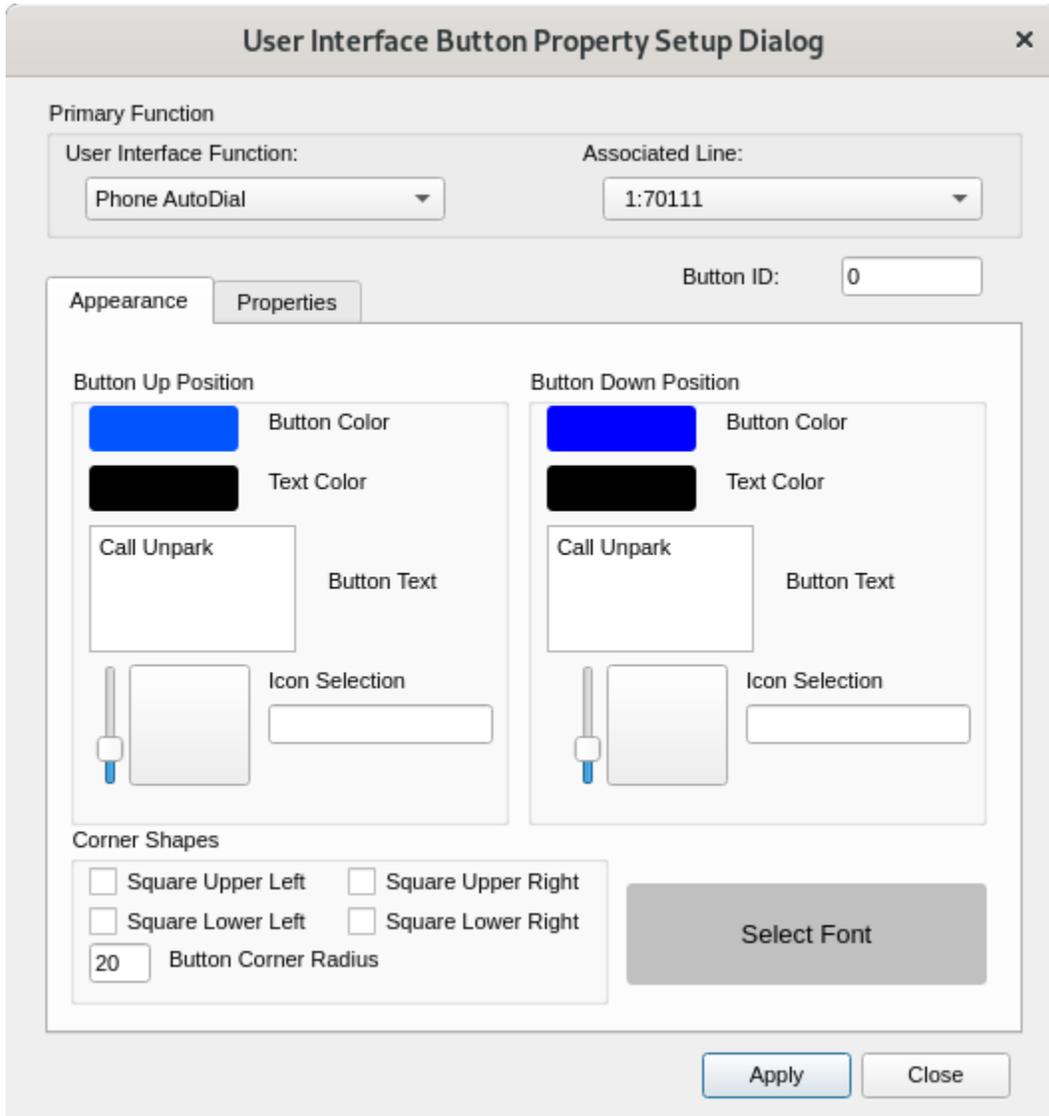
Appearance    **Properties**

Property	Value(s)	Unit(s)
1	Enable Autodial: <input checked="" type="checkbox"/>	
2	Number: <span style="border: 1px solid gray; padding: 2px;">*72</span>	
3	Transfer Blind: <input checked="" type="checkbox"/>	
4	Preset Dial String: <input type="checkbox"/>	
5	Popup Dialpad on Click: <input type="checkbox"/>	
6		
7		
8		
9		
10		
11		
12		
13		
14		
15		
16		

Apply
Close

### 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**.



The image shows a 'User Interface Button Property Setup Dialog' window. At the top, it has a title bar with a close button. Below the title bar, there are two dropdown menus: 'User Interface Function' set to 'Phone AutoDial' and 'Associated Line' set to '1:70111'. To the right of these is a 'Button ID' field with the value '0'. Below this, there are two tabs: 'Appearance' (selected) and 'Properties'. The 'Appearance' tab is divided into two columns: 'Button Up Position' and 'Button Down Position'. Each column has a 'Button Color' (blue), a 'Text Color' (black), a 'Button Text' field (containing 'Call Unpark'), and an 'Icon Selection' field with a slider and a selection box. At the bottom of the 'Appearance' tab, there are four checkboxes for 'Corner Shapes': 'Square Upper Left', 'Square Upper Right', 'Square Lower Left', and 'Square Lower Right'. Below these is a 'Button Corner Radius' field with the value '20'. To the right of the corner shapes is a 'Select Font' button. At the bottom of the dialog are 'Apply' and 'Close' buttons.

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**.

**User Interface Button Property Setup Dialog** ✕

Primary Function

User Interface Function: Phone AutoDial Associated Line: 1:70111

Button ID: 0

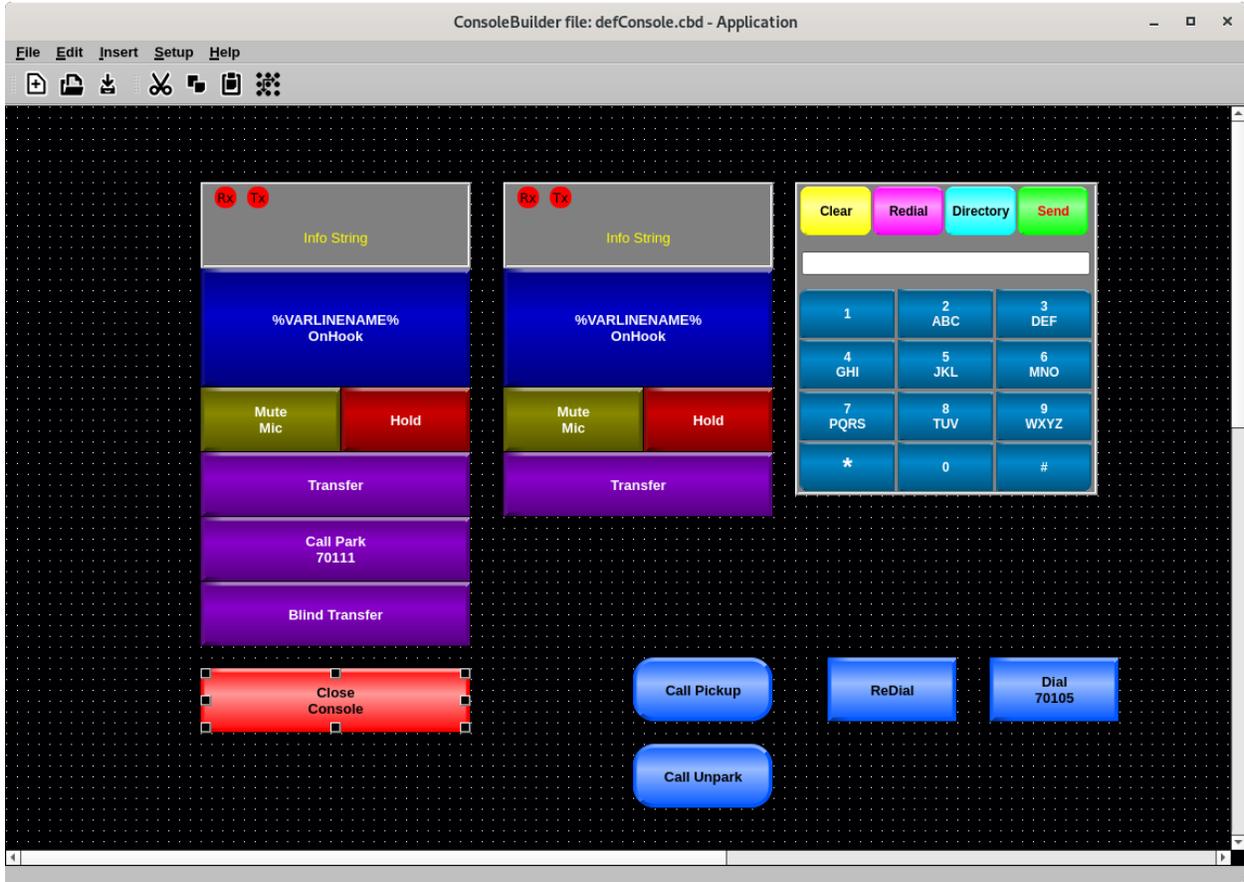
Appearance Properties

Property	Value(s)	Unit(s)
1	Dial String when Clicked: <span style="border: 1px solid gray; padding: 2px;">*71</span>	
2	Preset Dial String: <input type="checkbox"/>	
3	Preset String: <span style="border: 1px solid gray; padding: 2px;"> </span>	
4	Popup Dialpad on Click: <input type="checkbox"/>	
5	Dial on Associated Line: <input checked="" type="checkbox"/>	
6	Enable Autodial #2: <input checked="" type="checkbox"/>	
7	Enable Preset Dial #2: <input type="checkbox"/>	
8	Number 2: <span style="border: 1px solid gray; padding: 2px;">70111</span>	
9		
10		
11		
12		
13		
14		
15		
16		

Apply
Close

## 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/consulesuite/defConsole.cbd`. The layout used for Interoperability tests included a second extension.

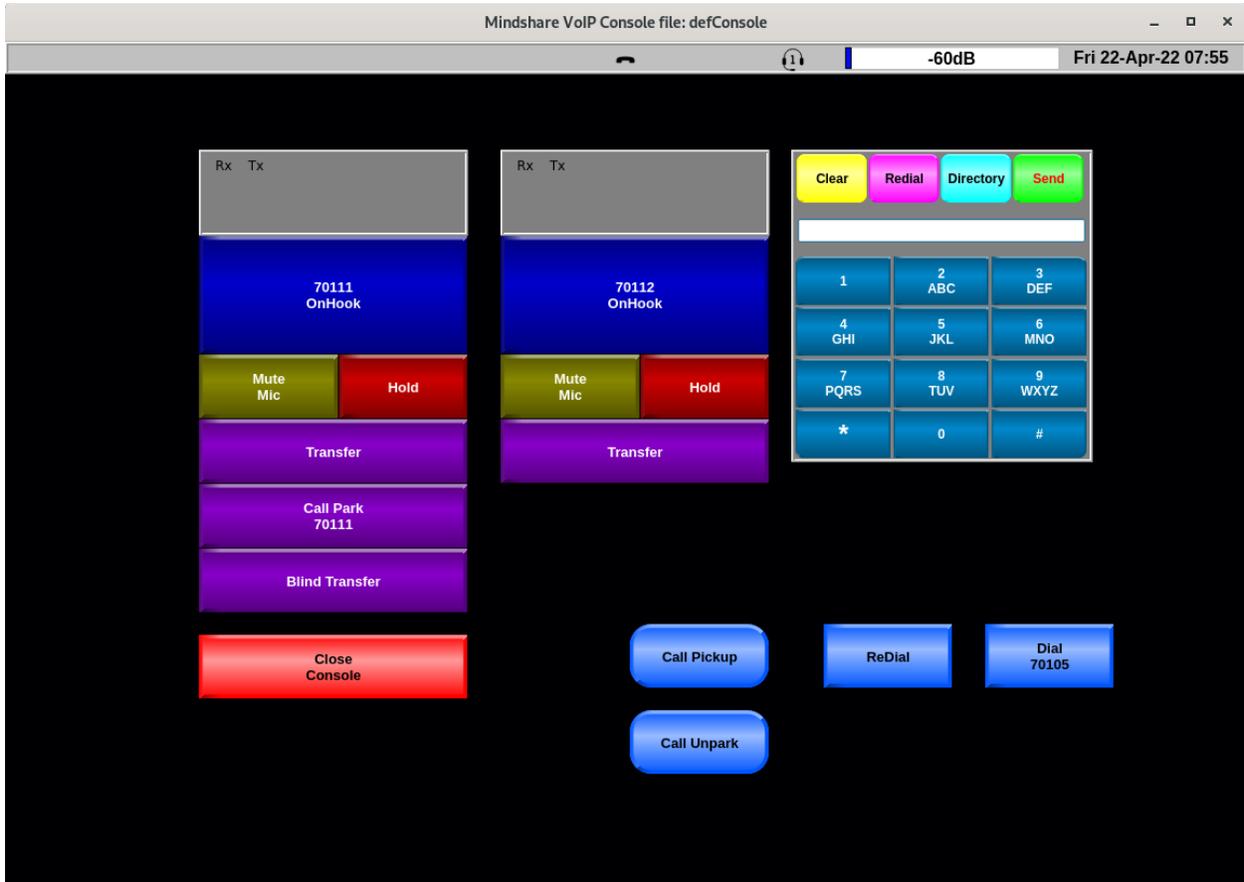


## 8. Verification Steps

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

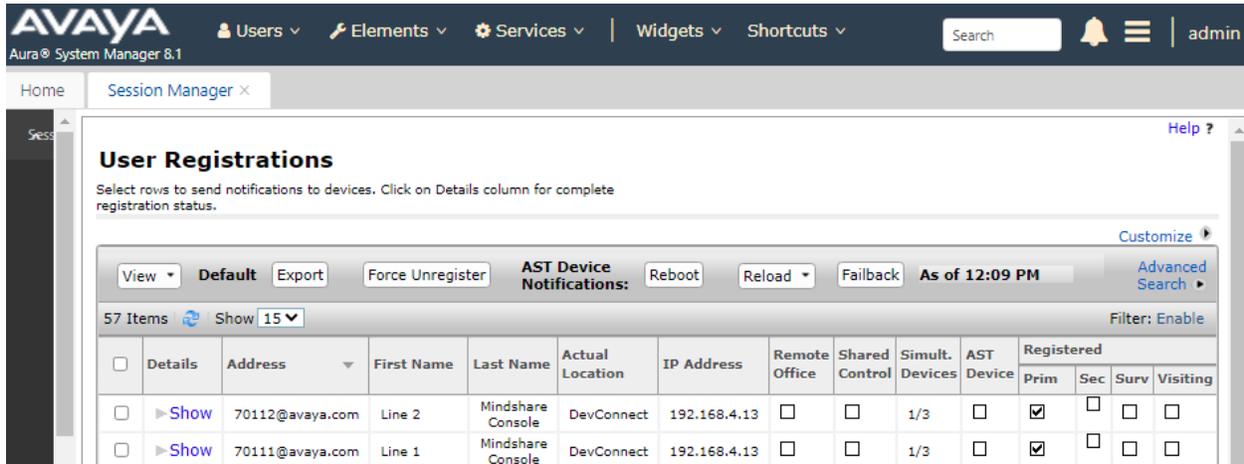
### 8.1. Launch Console

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.



The screenshot shows the Avaya Aura System Manager 8.1 interface. The main content area is titled "User Registrations" and contains a table with 57 items. The table has columns for "Details", "Address", "First Name", "Last Name", "Actual Location", "IP Address", "Remote Office", "Shared Control", "Simult. Devices", "AST Device", and "Registered". The "Registered" column has sub-columns for "Prim", "Sec", "Surv", and "Visiting". Two rows are visible, both with "Prim" checked and "AST Device" unchecked.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered			
											Prim	Sec	Surv	Visiting
<input type="checkbox"/>	<a href="#">Show</a>	70112@avaya.com	Line 2	Mindshare Console	DevConnect	192.168.4.13	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	<a href="#">Show</a>	70111@avaya.com	Line 1	Mindshare Console	DevConnect	192.168.4.13	<input type="checkbox"/>	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

## 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](https://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](https://customer.css-mindshare.com).

[3] *MS0101\_UM\_ConsoleApplicationManual*

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