



DevConnect Program

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 3.32.1 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. CSS Mindshare 100500 MaxPlus Dispatch Console incorporates telephony to integrate both radio and telephone functions. This solution also includes ConsoleBuilder 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 (ConsoleBuilder 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.
- 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.
- Incoming calls will not alert on MaxPlus Dispatch Console if an Avaya SIP phone is registered with the same SIP user using MDA (multi-device access).
- 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:

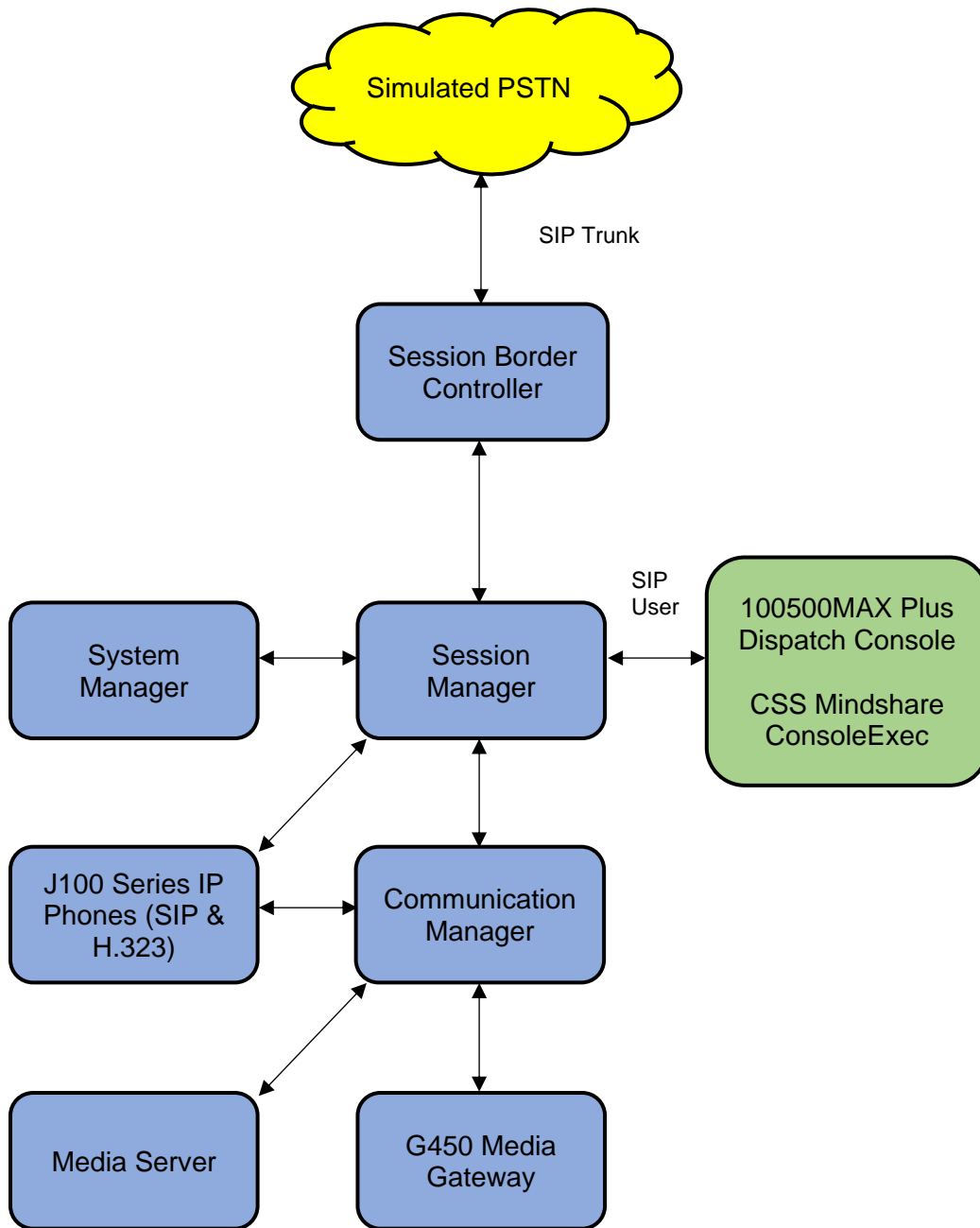


Figure 1: Avaya SIP Network with CSS Mindshare 100500 MaxPlus Dispatch Console

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	10.1.3.0.1 FP3P1 01.0.890.0-27893
Avaya Aura® Session Manager	10.1.3.0 FP3 10.1.3.0.1013007
Avaya Aura® System Manager	10.1.3.0 FP3 10.1.3.0.075713
Avaya Session Border Controller for Enterprise	10.1.2.0-64-23285
Avaya G450 Media Gateway	42.22.0
Avaya Aura® Media Server	10.1.0.147
Avaya J179 IP Phone	4.1.1.0.7 (SIP) 6.8.5.4 (H.323)
CSS Mindshare 100500 MaxPlus Dispatch Console	3.32.1 Debian GNU/Linux 10 (buster) Gnome 3.30.2
CSS Mindshare Console Suite (Includes ConsoleBuilder and ConsoleExec)	3.32.1

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: V20                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                         Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000    107
Maximum Stations: 150            72
Maximum XMOBILE Stations: 36000  0
Maximum Off-PBX Telephones - EC500: 150  0
Maximum Off-PBX Telephones - OPS: 150  41
Maximum Off-PBX Telephones - PBFMC: 150  0
Maximum Off-PBX Telephones - PVFMC: 150  0
Maximum Off-PBX Telephones - SCCAN: 0    0
Maximum Off-PBX Telephones - EMX: 150   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: 65535
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
                        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 SRTP: best-effort
1: 1-srtp-aescm128-hmac80
2: 10-srtp-aescm256-hmac80
3: 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.

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: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

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 10.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Routing' tab is selected. The 'SIP Entity Details' form for 'sm10' is displayed. The 'General' section contains the following fields:

- Name:** sm10
- IP Address:** 10.64.110.212
- SIP FQDN:**
- Type:** Session Manager
- Notes:**
- Location:** DevConnect
- Outbound Proxy:**
- Time Zone:** America/Denver
- Minimum TLS Version:** Use Global Setting
- Credential name:**

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the form.

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.

The screenshot shows the 'Listen Ports' section of the Avaya Aura System Manager 10.1 interface. It displays a table with 3 items. The table has columns for 'Listen Ports', 'Protocol', 'Default Domain', 'Endpoint', and 'Notes'. The 'Filter' is set to 'Enable'.

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

At the bottom, there is a 'Select' dropdown menu with options: 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 Avaya Aura System Manager 10.1 interface for adding a new user profile. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and various menu items like 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile icon are also present. The main content area is titled 'User Profile | Add' and features a sidebar with tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Basic Info' tab is currently selected. The form fields are organized into two columns. The left column includes 'User Provisioning Rule' (a dropdown menu), 'Address', 'LocalizedName', and a 'Basic Info' section with fields for 'Last Name', 'First Name', 'Login Name', 'Description', 'Password', and 'Confirm Password'. The right column includes fields for 'Last Name (in Latin alphabet characters)', 'First Name (in Latin alphabet characters)', 'Middle Name', 'Email Address', 'User Type' (a dropdown menu), and 'Localized Display Name'. The 'Last Name' field is populated with 'Mindshare Console', the 'First Name' field with 'Line 1', and the 'Login Name' field with '70111@avaya.com'. The 'User Type' dropdown is set to 'Basic'.

6.3.2. Communication Address

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

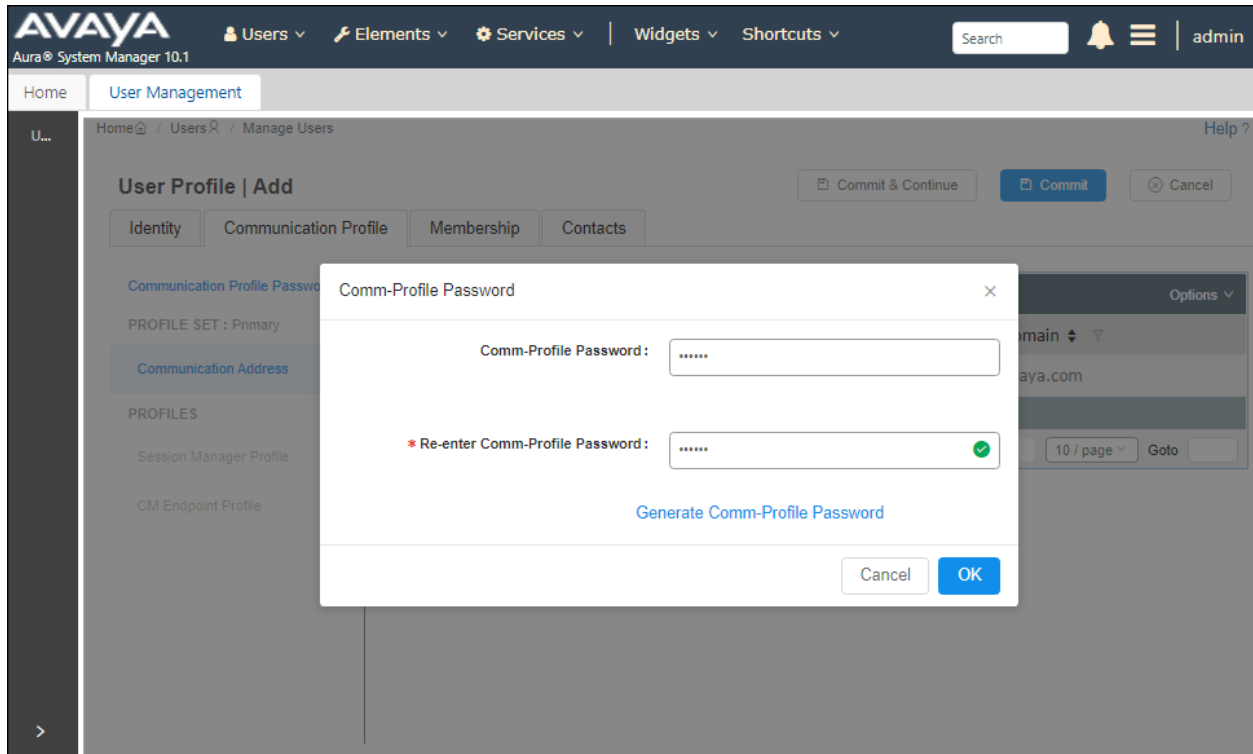
Enter the following attributes for the **Communication Address**:

- **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**).

The screenshot displays the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and tabs for 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also visible. The main content area is titled 'User Management' and shows a 'User Profile | Add' form. The 'Communication Profile' tab is selected, and the 'Communication Address' option is highlighted in the left sidebar. A modal dialog box titled 'Communication Address Add/Edit' is open, featuring a dropdown for '*Type:' set to 'Avaya SIP', and input fields for '*Fully Qualified Address:' with '70111' and a domain dropdown set to 'avaya.com'. The dialog has 'Cancel' and 'OK' buttons at the bottom.

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**.

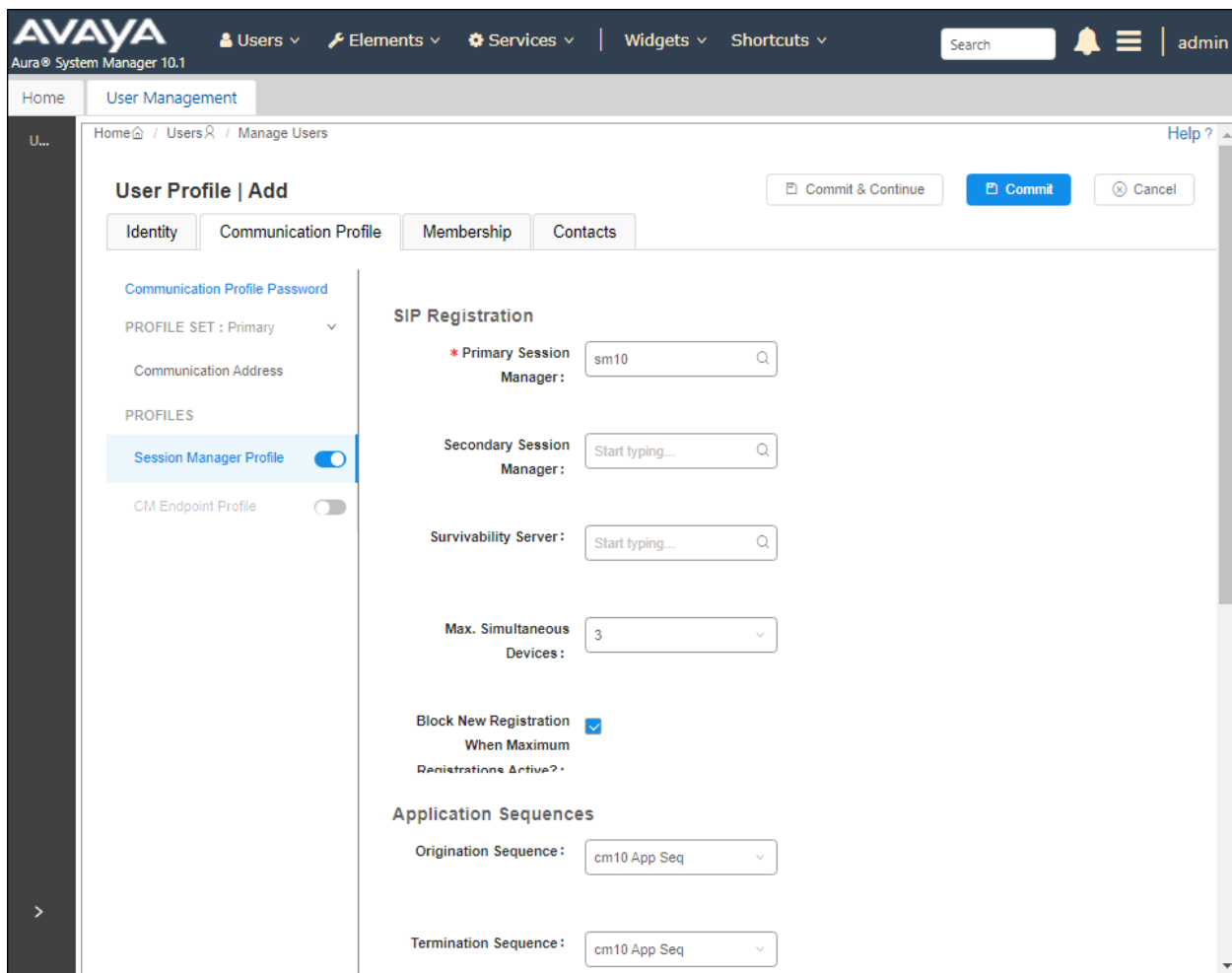


The screenshot displays the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and various menu items like 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and user profile 'admin' are also visible. The main content area is titled 'User Profile | Add' and has tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active. On the left sidebar, 'Communication Profile Password' is selected under the 'PROFILES' section. A modal window titled 'Comm-Profile Password' is open in the center, featuring two password input fields: 'Comm-Profile Password' and 'Re-enter Comm-Profile Password'. The 'Re-enter' field has a green checkmark icon indicating a match. Below the fields is a link 'Generate Comm-Profile Password' and 'Cancel'/'OK' buttons. The background shows the 'Communication Address' section with a dropdown menu and pagination controls.

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 displays the Avaya Aura System Manager 10.1 User Management 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 tabs for 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing a 'Communication Profile Password' section with a dropdown for 'PROFILE SET : Primary' and a 'Communication Address' field. Below this, the 'PROFILES' section contains a 'Session Manager Profile' toggle (which is turned on) and a 'CM Endpoint Profile' toggle (which is turned off). The 'SIP Registration' section includes a 'Primary Session Manager' field with the value 'sm10', a 'Secondary Session Manager' field with the placeholder 'Start typing...', a 'Survivability Server' field with the placeholder 'Start typing...', a 'Max. Simultaneous Devices' dropdown set to '3', and a checked checkbox for 'Block New Registration When Maximum Registrations Active?'. The 'Application Sequences' section at the bottom includes an 'Origination Sequence' dropdown set to 'cm10 App Seq' and a 'Termination Sequence' dropdown set to 'cm10 App Seq'.

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., **cm10**).
- **Profile Type:** Select **Endpoint**.
- **Template:** Select **J179_DEFAULT_CM_10_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 10.1 User Management interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and tabs for Users, Elements, Services, Widgets, and Shortcuts. A search bar and user profile (admin) are also visible. The main content area is titled 'User Profile | Add' and features tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active, showing a form for adding a new profile. The form includes fields for System (cm10), Profile Type (Endpoint), Extension (70111), Template (J179_DEFAULT_CM_10_1), Set Type (J179), Security Code, Port (IP), Voice Mail Number, Preferred Handle (Select), SIP URI (Select), and Sip Trunk. There are also checkboxes for 'Use Existing Endpoints', 'Calculate Route Pattern', 'Override Endpoint Name and Localized Name', and 'Delete on Unassign from User or on Delete User'. A 'Commit & Continue' button, a 'Commit' button, and a 'Cancel' button are located at the top right of the form area.

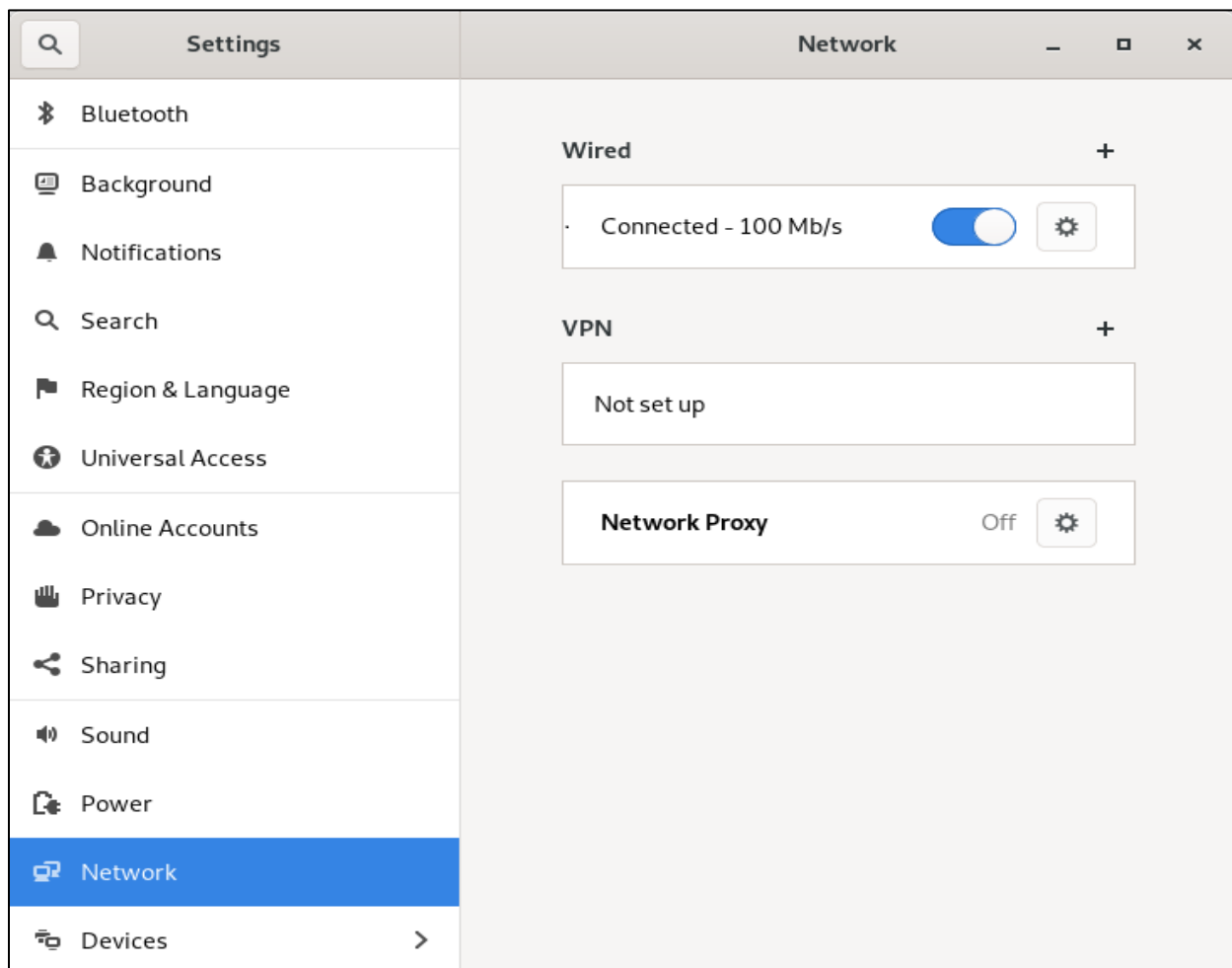
7. Configure CSS Mindshare 100500 MaxPlus Dispatch Console

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

- Configure IP Address
- Launch ConsoleBuilder
- 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

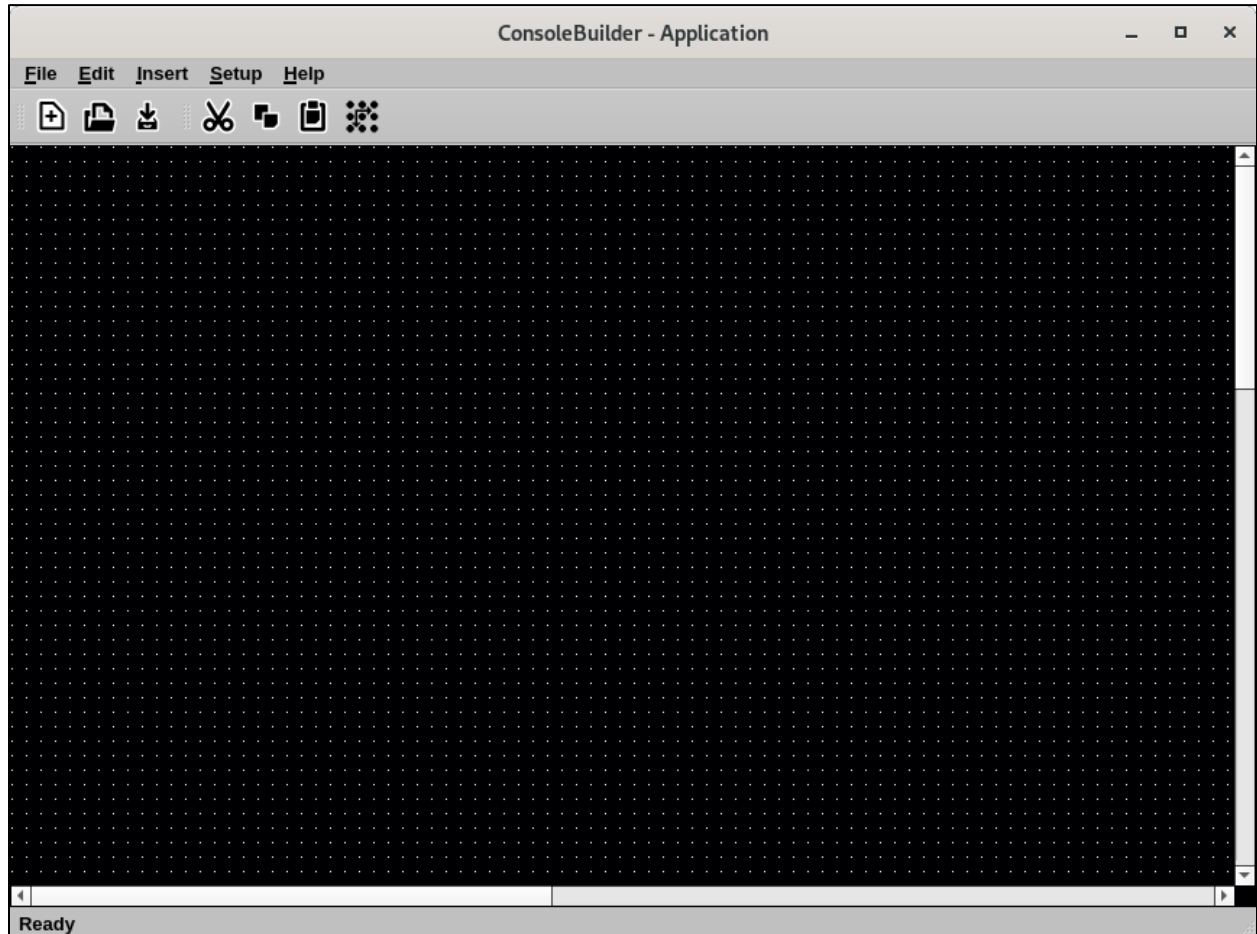
Metric

×

☐ Use this connection only for resources on its network

7.2. Launch ConsoleBuilder

The Console user interface is configured using ConsoleBuilder, 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:20hops

SIP Max Retry Count:3

SIP Time Before Retry:200ms

SIP Registration Time:1800sec

SIP Local Port Number:5060

☒ Auto Hold

☒ TCP

Phone Line Tone Control Parameters

Guard Tone Frequency:2175Hz

Function F1 Frequency:1950Hz

Guard Tone Level:0dB

Function F2 Frequency:1850Hz

Guard Tone Duration:130ms

Function Tone Level:-10dB

Hold Tone Frequency:2175Hz

Function Tone Duration:40ms

Hold Tone Level:-20dB

Radio Tone Burst Interval:7sec

Dial Tone Level:0dB

Phone Line Crosspatch VOX Parameters

VOX Trigger Level:-20dB

VOX Hangtime:3000ms

DTMF Signaling Parameters

DTMF Digit On Time:100ms

DTMF Flywheel:2000ms

DTMF Digit Off Time:100ms

DTMF Level:-10dB

DTMF Wait/Pause Time:500ms

RFC 2833 Flash Duration:1250ms

Phone Line Ringer Levels

All Lines OnHook:

One or more lines Offhook:

Ring Level:-8dB

Ring Level:-14dB

Speaker (1-8):1

Speaker (1-8):2

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

Digit Map:

Cancel

OK

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:

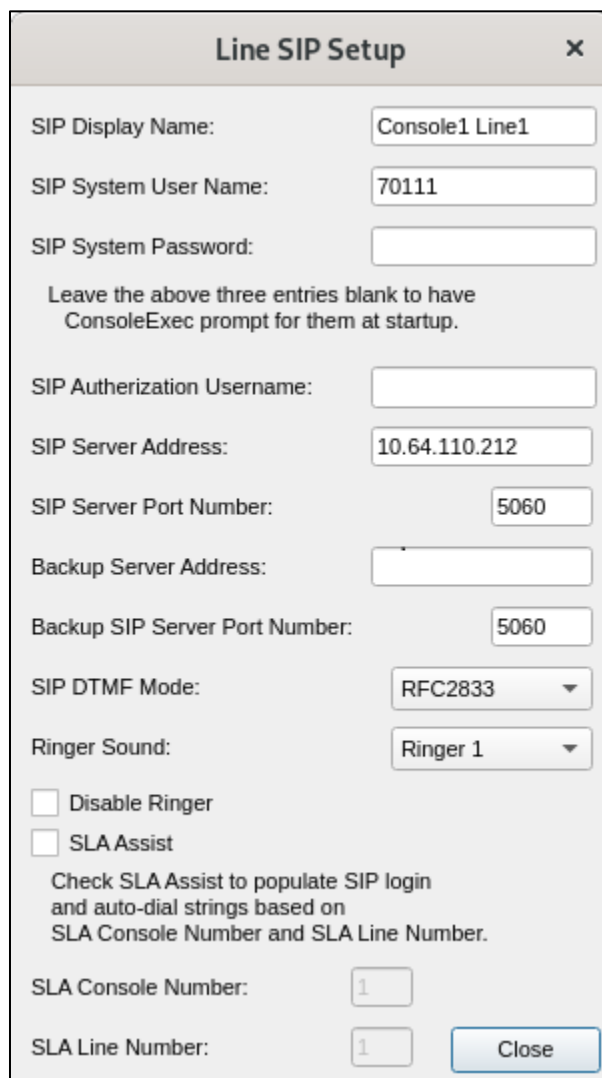
AutoFill

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 a title bar containing a close button (X). The dialog contains several input fields and checkboxes. The 'SIP Display Name' field is filled with 'Console1 Line1'. The 'SIP System User Name' field is filled with '70111'. The 'SIP System Password' field is empty. Below these fields is a note: 'Leave the above three entries blank to have ConsoleExec prompt for them at startup.' The 'SIP Authorization Username' field is empty. The 'SIP Server Address' field is filled with '10.64.110.212'. The 'SIP Server Port Number' field is filled with '5060'. The 'Backup Server Address' field is empty. The 'Backup SIP Server Port Number' field is filled with '5060'. The 'SIP DTMF Mode' dropdown is set to 'RFC2833'. The 'Ringer Sound' dropdown is set to 'Ringer 1'. There are two checkboxes: 'Disable Ringer' and 'SLA Assist', both of which are unchecked. Below these is a note: 'Check SLA Assist to populate SIP login and auto-dial strings based on SLA Console Number and SLA Line Number.' The 'SLA Console Number' field is filled with '1'. The 'SLA Line Number' field is filled with '1'. A 'Close' button is located at the bottom right of the dialog.

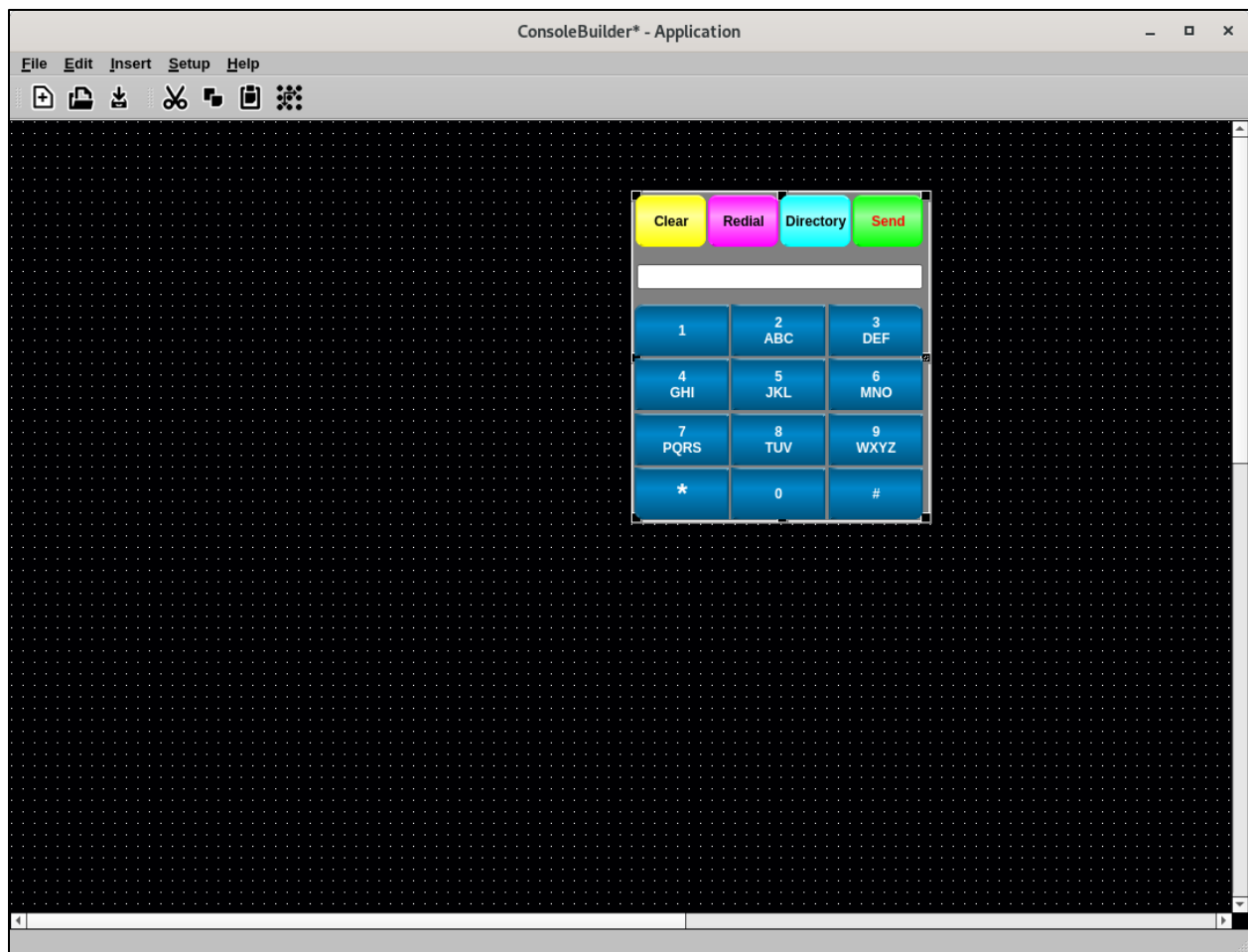
Line SIP Setup	
SIP Display Name:	Console1 Line1
SIP System User Name:	70111
SIP System Password:	
Leave the above three entries blank to have ConsoleExec prompt for them at startup.	
SIP Authorization Username:	
SIP Server Address:	10.64.110.212
SIP Server Port Number:	5060
Backup Server Address:	
Backup 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
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. ConsoleBuilder 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

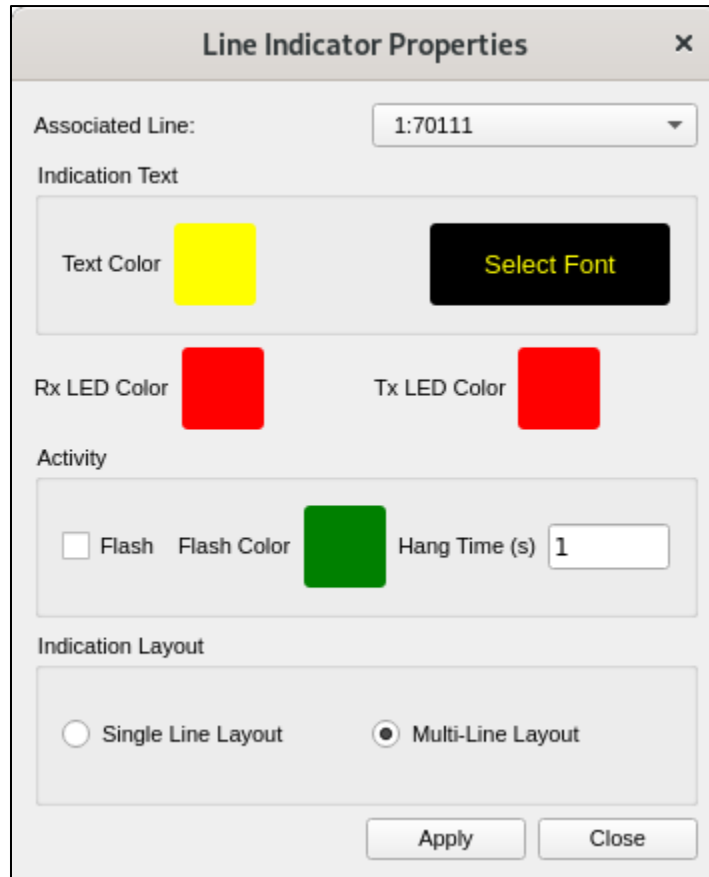
7.5.1. Dial Keypad

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**.



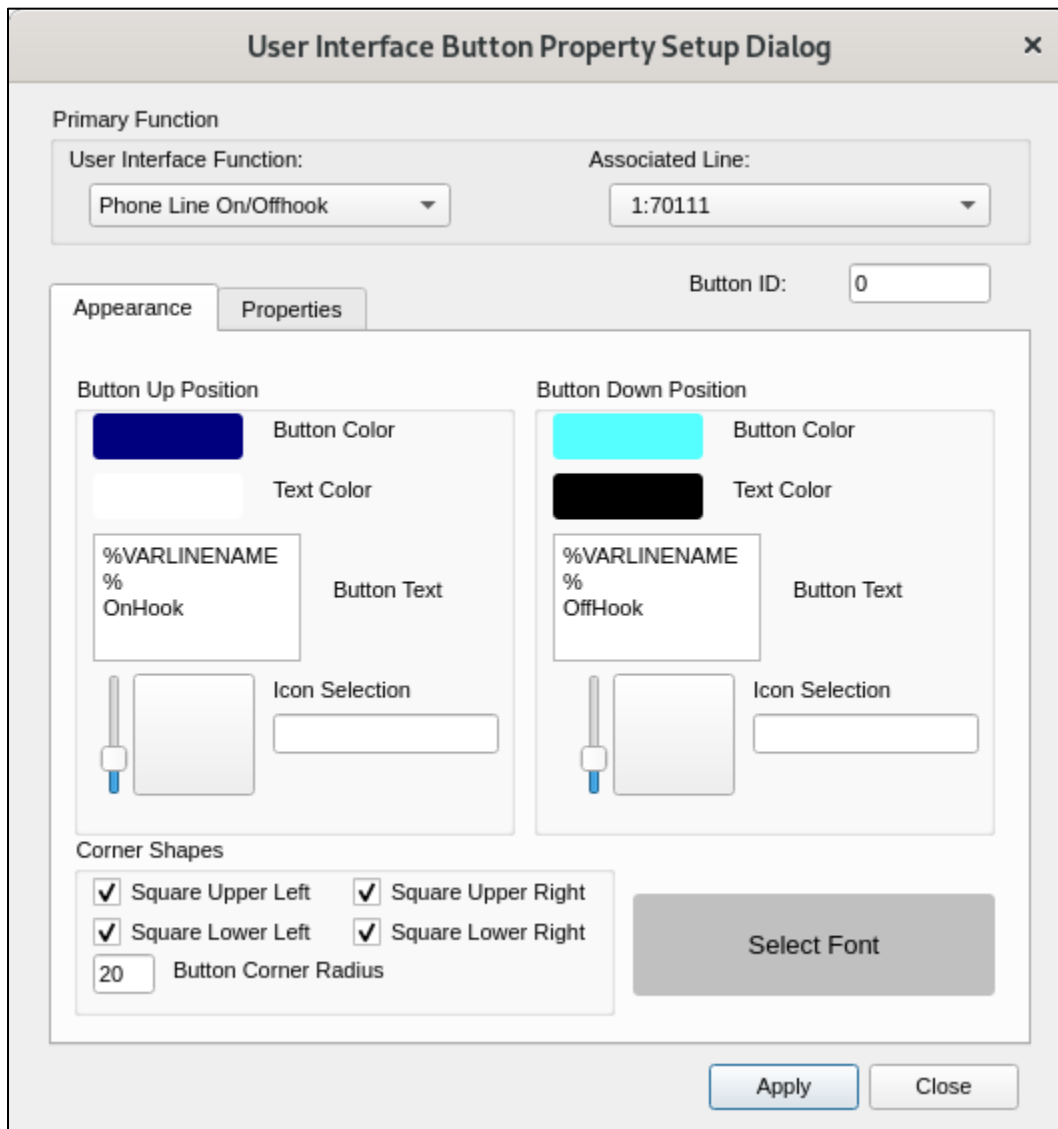
The image shows a 'Line Indicator Properties' dialog box with a close button (X) in the top right corner. The dialog is organized into several sections:

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

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 dialog box is titled "User Interface Button Property Setup Dialog" and features a close button (X) in the top right corner. It is divided into two main sections: "Primary Function" and "Appearance/Properties".

Primary Function:

- User Interface Function:** A dropdown menu set to "Phone Line On/Offhook".
- Associated Line:** A dropdown menu set to "1:70111".

Appearance/Properties:

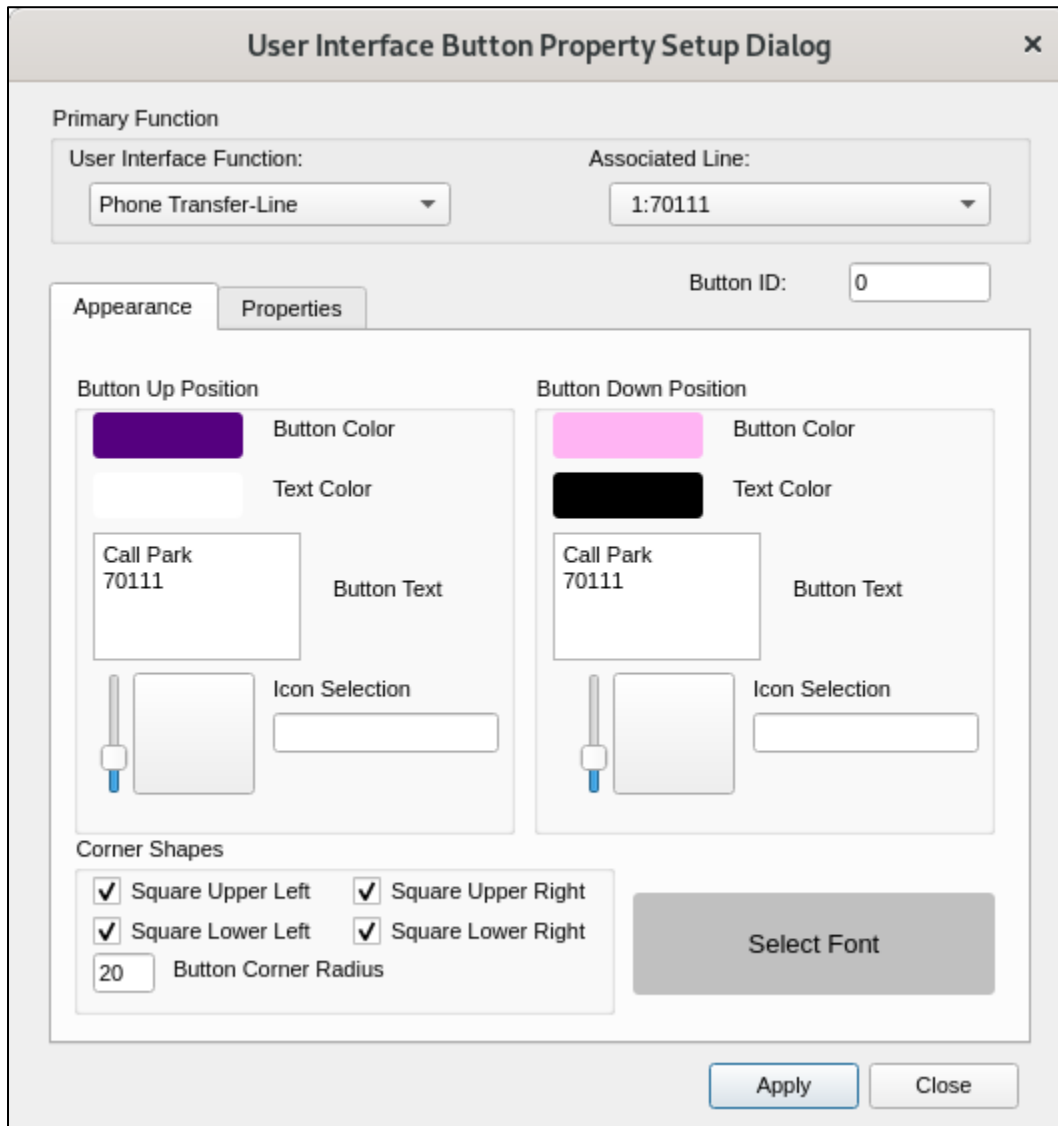
- Button ID:** A text field containing "0".
- Appearance Tab:** Contains settings for the button's visual state.
 - Button Up Position:**
 - Button Color:** A color swatch showing dark blue.
 - Text Color:** A color swatch showing white.
 - Button Text:** A text field containing "%VARLINENAME% OnHook".
 - Icon Selection:** A vertical slider and a square icon placeholder.
 - Button Down Position:**
 - Button Color:** A color swatch showing cyan.
 - Text Color:** A color swatch showing black.
 - Button Text:** A text field containing "%VARLINENAME% OffHook".
 - Icon Selection:** A vertical slider and a square icon placeholder.
- Corner Shapes:**
 - Four checkboxes: "Square Upper Left", "Square Upper Right", "Square Lower Left", and "Square Lower Right", all of which are checked.
 - Button Corner Radius:** A text field containing "20".
- Select Font:** A large grey button.

Buttons: "Apply" and "Close" buttons are located at the bottom right of the dialog.

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.

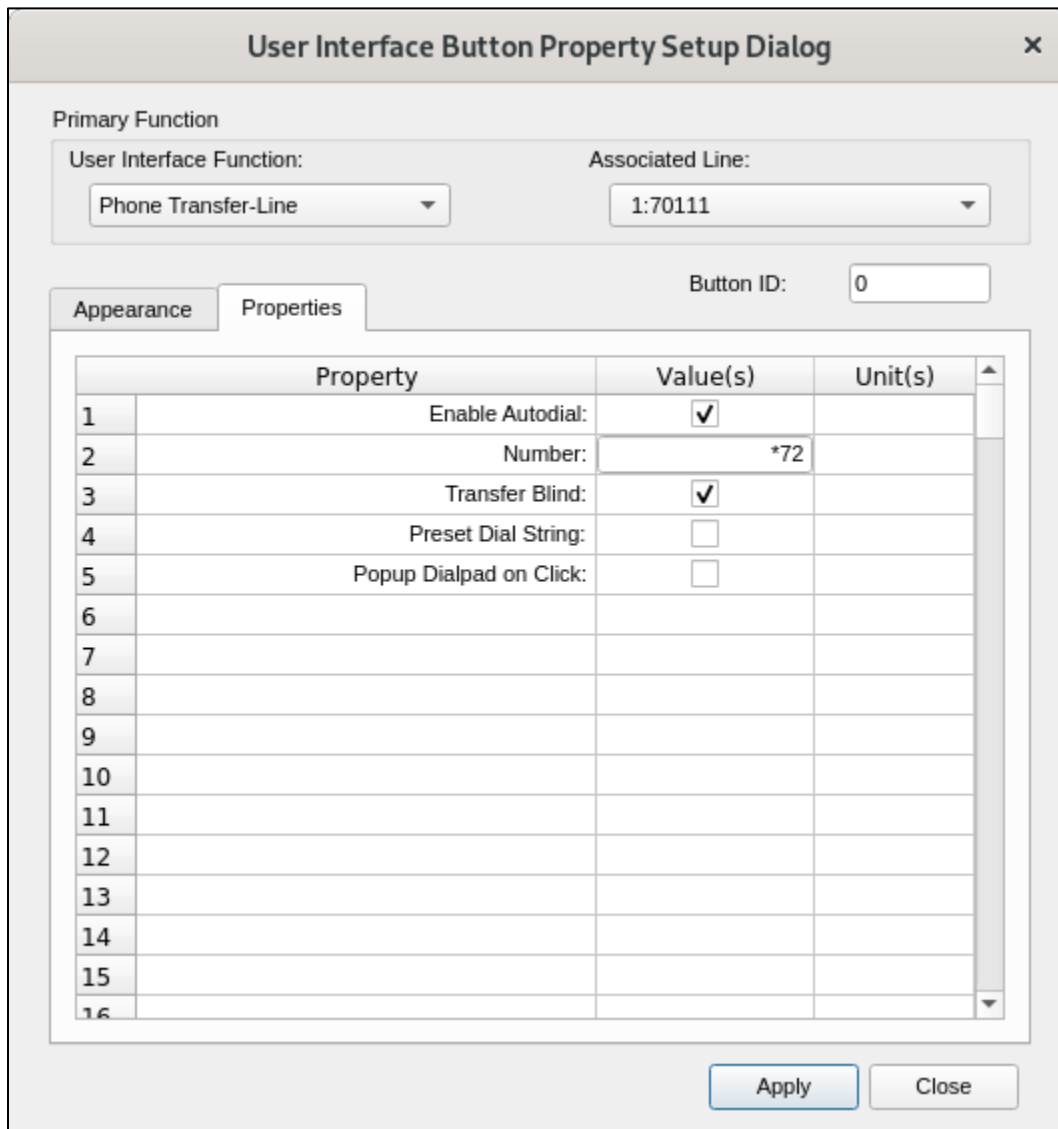
Select **Insert → Insert Button Control**. Adjust the size and position of the button control on the grid. Mouse over the new button and right click on **Properties**. Select the Line defined in **Section 7.3** for **Associated Line**, e.g., **70111**. Select **Phone Transfer-Line** for **User Interface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



The dialog box is titled "User Interface Button Property Setup Dialog" and contains the following sections:

- Primary Function:**
 - User Interface Function:** Phone Transfer-Line (dropdown)
 - Associated Line:** 1:70111 (dropdown)
- Buttons:** Appearance (selected) and Properties
- Button ID:** 0 (text field)
- Button Up Position:**
 - Button Color:** Purple color swatch
 - Text Color:** White color swatch
 - Button Text:** Call Park 70111 (text field)
 - Icon Selection:** Includes a slider and an empty text field.
- Button Down Position:**
 - Button Color:** Pink color swatch
 - Text Color:** Black color swatch
 - Button Text:** Call Park 70111 (text field)
 - Icon Selection:** Includes a slider and an empty text field.
- Corner Shapes:**
 - ☒ Square Upper Left
 - ☒ Square Upper Right
 - ☒ Square Lower Left
 - ☒ Square Lower Right
 - Button Corner Radius:** 20 (text field)
- Select Font:** A grey button.
- Buttons:** Apply and Close

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**.



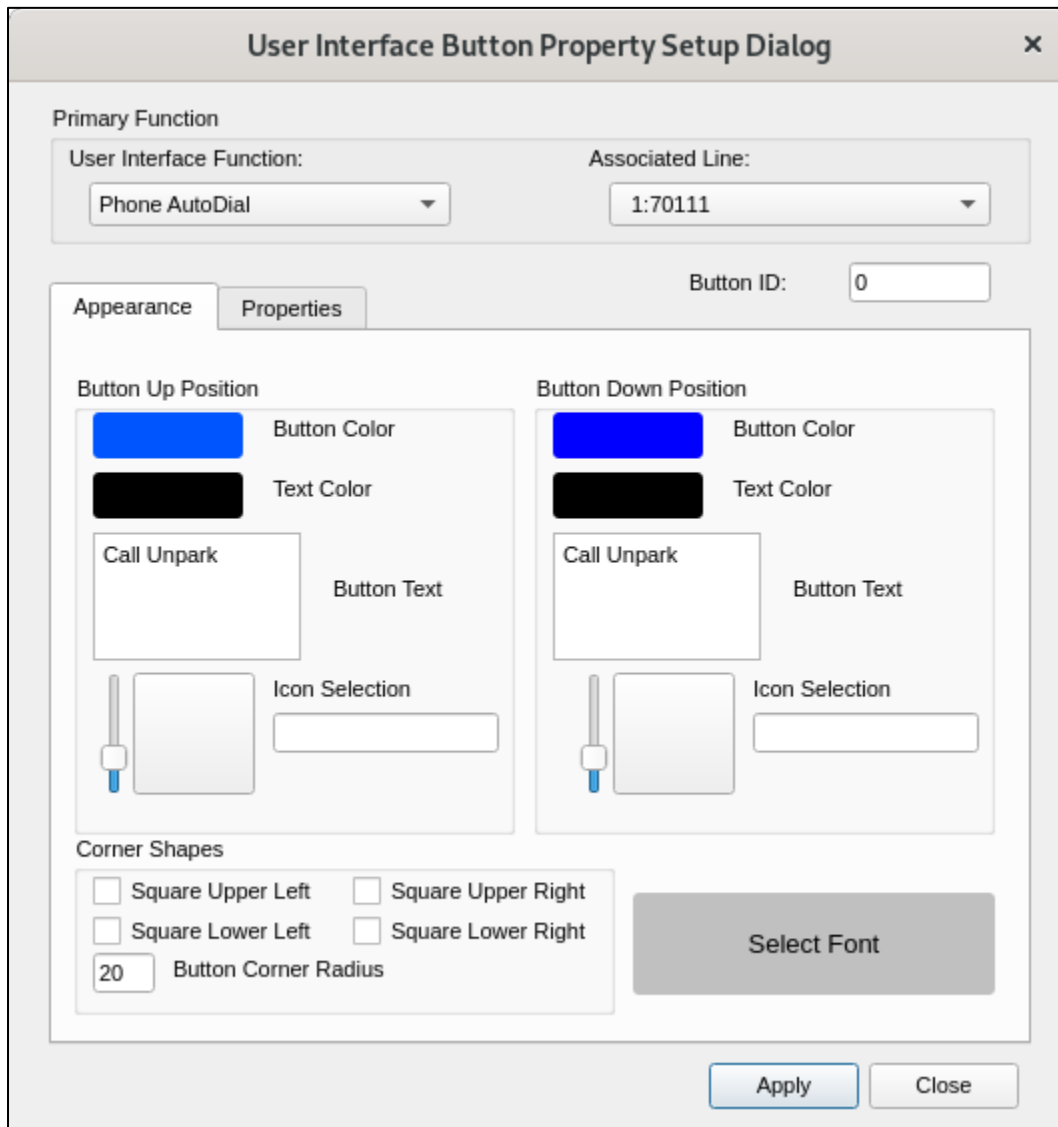
The dialog box is titled "User Interface Button Property Setup Dialog" and has a close button (X) in the top right corner. It contains a "Primary Function" section with two dropdown menus: "User Interface Function:" (set to "Phone Transfer-Line") and "Associated Line:" (set to "1:70111"). Below this is a "Button ID:" field with the value "0". The dialog has two tabs: "Appearance" and "Properties", with "Properties" currently selected. The "Properties" tab contains a table with 16 rows and 3 columns: "Property", "Value(s)", and "Unit(s)". The table is populated with the following data:

	Property	Value(s)	Unit(s)
1	Enable Autodial:	<input checked="" type="checkbox"/>	
2	Number:	*72	
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			

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

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 dialog box is titled "User Interface Button Property Setup Dialog" and features a close button (X) in the top right corner. It is divided into two main sections: "Primary Function" and "Appearance".

Primary Function:

- User Interface Function:** A dropdown menu with "Phone AutoDial" selected.
- Associated Line:** A dropdown menu with "1:70111" selected.
- Button ID:** A text field containing "0".

Appearance:

The "Appearance" section has two tabs: "Appearance" (selected) and "Properties".

Button Up Position:

- Button Color:** A blue color swatch.
- Text Color:** A black color swatch.
- Button Text:** A text field containing "Call Unpark".
- Icon Selection:** A vertical slider and a square icon selection area.

Button Down Position:

- Button Color:** A blue color swatch.
- Text Color:** A black color swatch.
- Button Text:** A text field containing "Call Unpark".
- Icon Selection:** A vertical slider and a square icon selection area.

Corner Shapes:

- ☐ Square Upper Left
- ☐ Square Upper Right
- ☐ Square Lower Left
- ☐ Square Lower Right
- Button Corner Radius:** A text field containing "20".

Select Font: A button labeled "Select Font".

Buttons: "Apply" and "Close" buttons are located at the bottom right of the dialog.

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:

Associated Line:

Button ID:

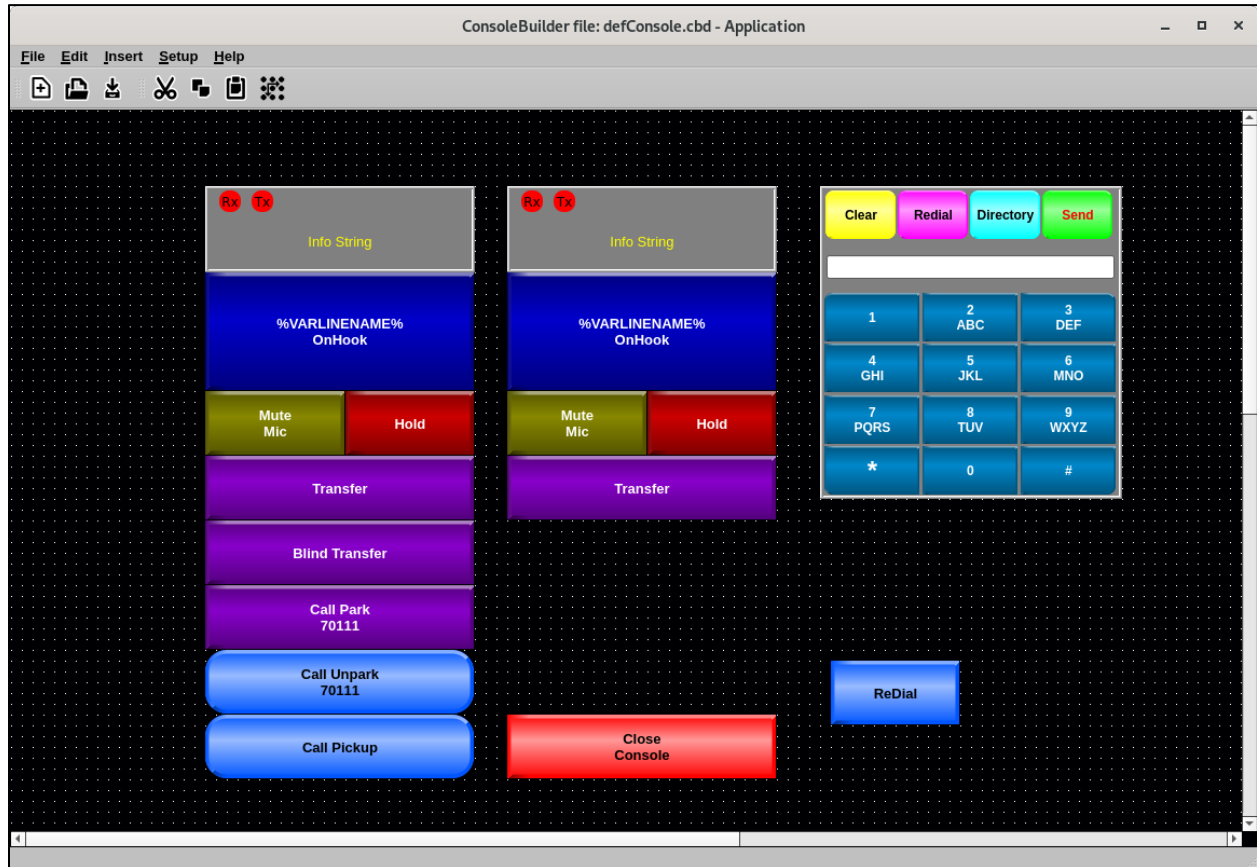
Appearance

Properties

Property	Value(s)	Unit(s)
1	Dial String when Clicked: <input type="text" value="*71"/>	
2	Preset Dial String: <input type="checkbox"/>	
3	Preset String: <input type="text"/>	
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: <input type="text" value="70111"/>	
9		
10		
11		
12		
13		
14		
15		
16		

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.

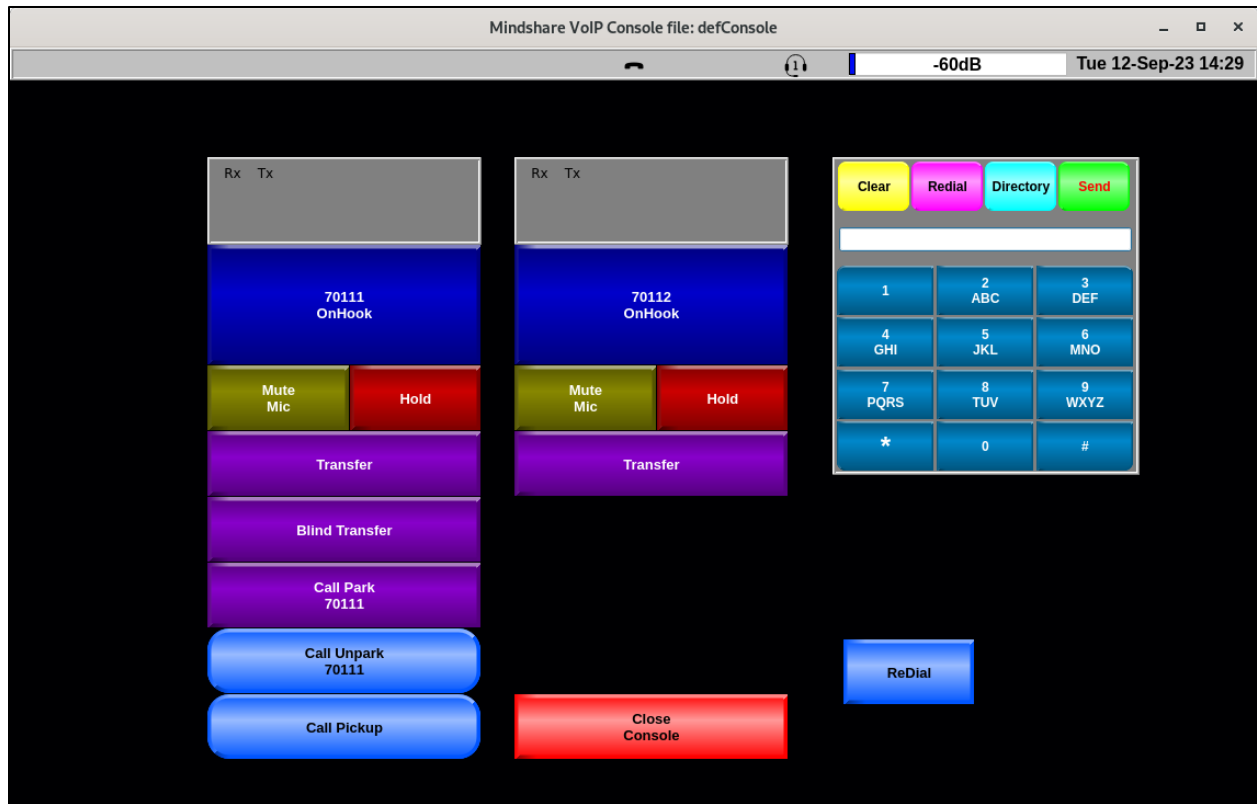


8. Verification Steps

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 ConsoleExec

Launch the ConsoleExec application from the Administrator account on the system through the **Applications→Mindshare→ConsoleExec** 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.

AVAYA

Aura® System Manager 10.1

Users

Elements

Services

Widgets

Shortcuts

Search

admin

Home

Session Manager

S...

Help

User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

View

Default

Export

Force Unregister

AST Device Notifications:

Reboot

Reload

Failback

As of 2:33 PM

Customize

Advanced Search

7 Items

Show

All

Filter: Enable

<input type="checkbox"/>	Details	Address	First Name	Last Name	Actual Location	IP Address	Policy	Shared Control	Simult. Devices	AST Device	Registered					
											Prim	Sec	3rd	4th	Surv	Visiting
<input type="checkbox"/>	Show	70112@avaya.com	Line 2	Mindshare Console	DevConnect	192.168.5.105	fixed	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	70111@avaya.com	Line 1	Mindshare Console	DevConnect	192.168.5.105	fixed	<input type="checkbox"/>	1/3	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input 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 3.32.1 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1. 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 2.2**.

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 6, Release 10.1.x, June 2023
- [2] *Administering Avaya Aura® Session Manager*, Issue 6, Release 10.1.x, May 2023

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

- [3] *MS0101_UM_ConsoleApplicationManual*, Version 1.15, June 23,2022

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