



Avaya Solution & Interoperability Test Lab

Application Notes for CSS Mindshare 100500 MaxPlus Dispatch Console integration with Avaya IP Office 11.1 - 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 IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. 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 3.27.2 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. 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 IP Office. 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/resume, mute, and transfer. MaxPlus Dispatch does not support conferencing. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using IP Office Short Codes. 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 IP Office.
- 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/resume, mute, redial, and blind and attended call transfer.
- Extended telephony features using IP Office short codes for Call Forwarding, Call Park/Unpark, and Call Pickup All.
- Use of programmable buttons (Console Builder button controls) for Call Pickup All 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 not supported.
- Call on Hold Reminder is not supported.
- Audio tones for invalid numbers or outbound call screening are not given but MaxPlus Dispatch Console line indicator display notifications are made.
- MaxPlus Dispatch Console line indicator display does not show called parties. In most cases, MaxPlus Dispatch 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 a call is answered, MaxPlus Dispatch Console line indicator display may show the calling party. Variations occur among the type of endpoints calling and whether they are registered to IP Office Server Edition or IP Office 500 V2 Expansion System.
- Calling the IP Office voicemail system via IP Office default short code *17 is not reliable. This is addressed in future MaxPlus Dispatch Console release 3.28.3.
- Calls cannot be forwarded on busy/ring no answer/forward unconditional to MaxPlus Dispatch Console. This is addressed in future MaxPlus Dispatch Console release 3.28.4.

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:

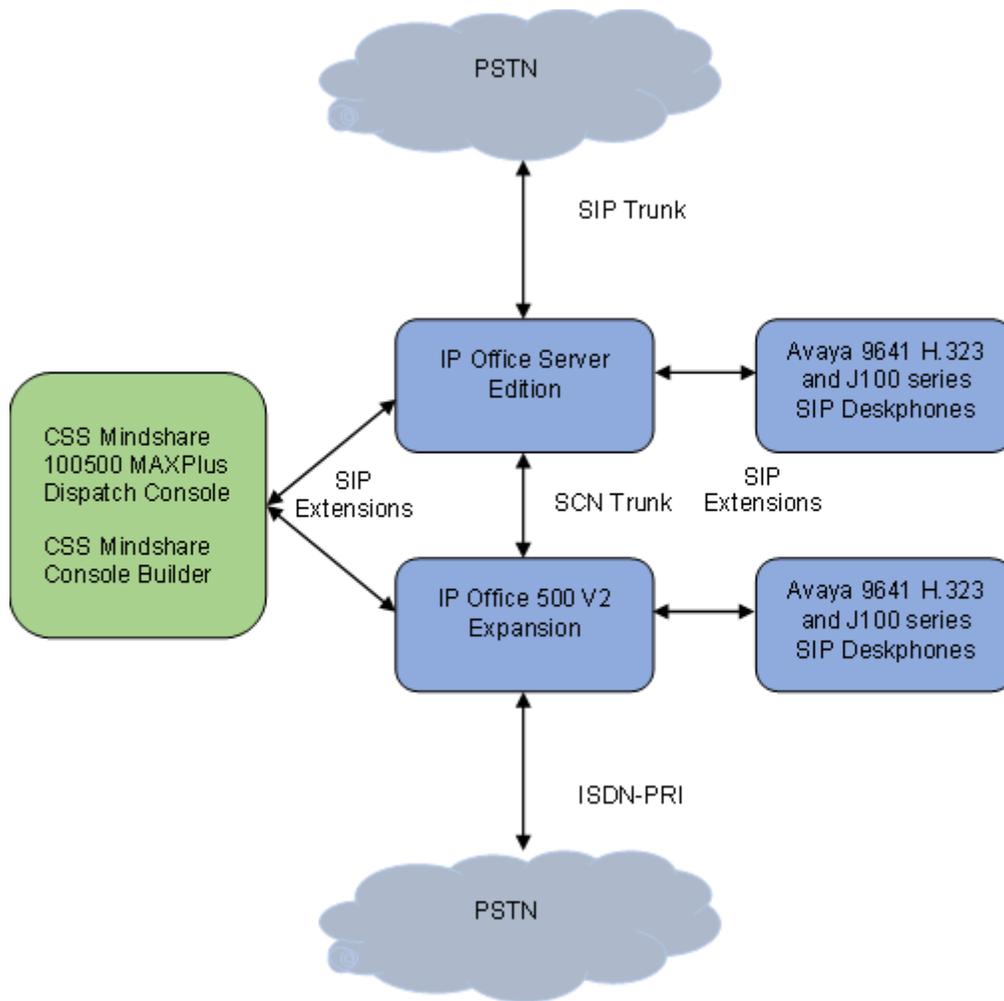


Figure 1: Avaya SIP Network with CSS Mindshare 100500 MaxPlus Dispatch Console

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office 500 V2 Expansion	11.1.2.2.0
Avaya IP Office Server Edition	11.1.2.2.0
Avaya 9641G IP Deskphone	6.8.3.0.4 (H.323)
Avaya J179 IP Phone	4.0.7.0.7 (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

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office Server Edition

This section provides the procedures for configuring Avaya IP Office Server Edition. The procedures include the following areas:

- Verify License
- Obtain LAN IP Address
- Administer SIP Registrar
- Administer IP Codecs
- Administer SIP Extension for MaxPlus Dispatch Console
- Administer SIP User for MaxPlus Dispatch Console

Note: This section covers the configuration of Avaya IP Office Server Edition, but the configuration is the same for Avaya IP Office 500 V2 Expansion System.

5.1. Verify License

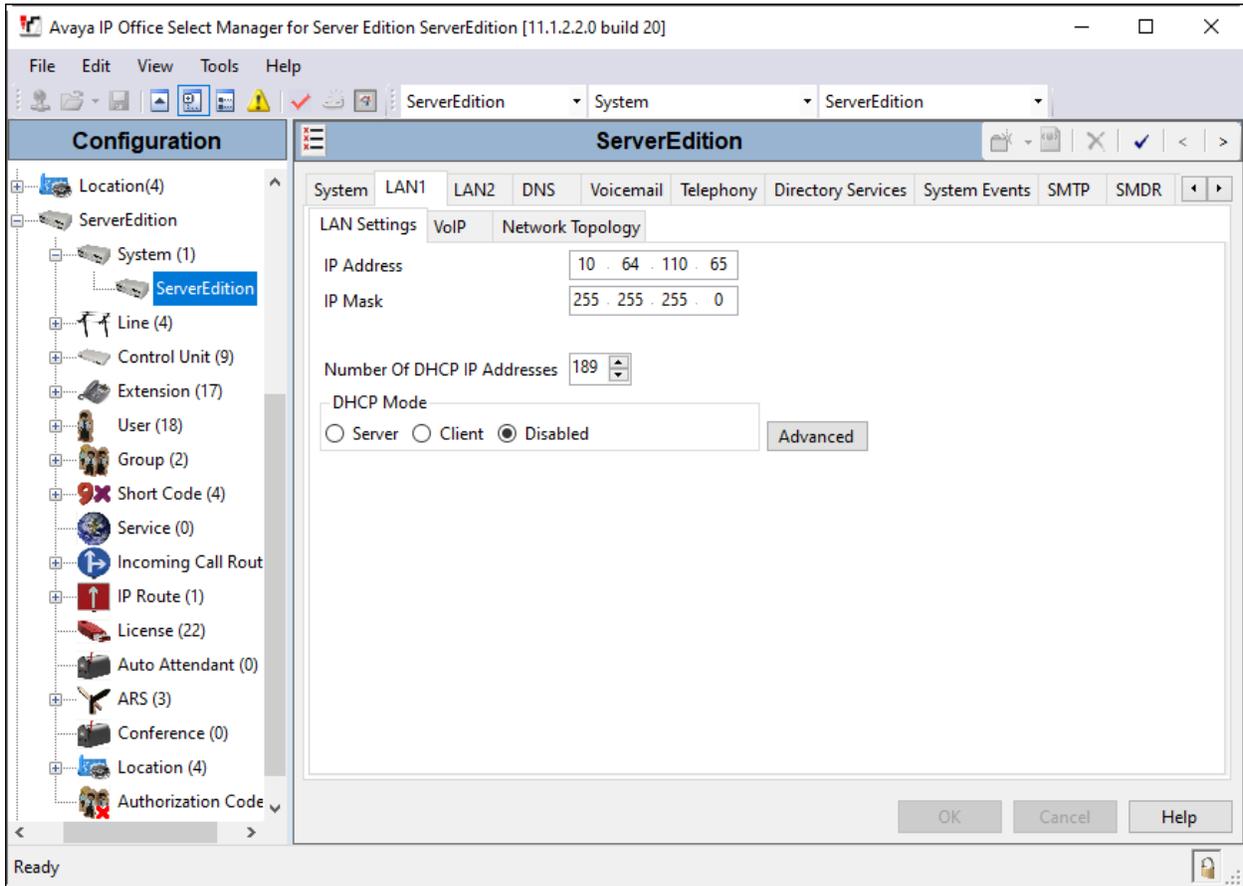
From a PC with **IP Office Admin Suite** installed, invoke **IP Office Manager**. Select the proper primary IP Office system (not shown), and log in using the appropriate credentials. Avaya IP Office Manager for Server Edition screen is displayed. From the configuration tree in the left pane, select **License** under the IP Office system that will be used to display a list of licenses in the right pane. Verify that there are sufficient licenses for **3rd Party IP Endpoints**.

The screenshot shows the Avaya IP Office Select Manager for Server Edition interface. The left pane displays a configuration tree with the following items: Location(4), ServerEdition, System (1), Line (4), Control Unit, Extension (1), User (18), Group (2), Short Code (1), Service (0), Incoming Call, IP Route (1), License (22), Auto Attend, ARS (3), Conference, Location (4), Authorizatio, and IP500v2. The 'License (22)' item is selected and highlighted in blue. The right pane displays a table of licenses with the following columns: Feature, Instances, Status, Expiration Date, and Source. The '3rd Party IP Endpoints' license is highlighted in blue.

Feature	Instances	Status	Expiration Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPPro Recordings Administrators	1	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	1	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	100	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal
Wave User	16	Obsolete	Never	PLDS Nodal
3rd Party IP Endpoints	1000	Valid	Never	PLDS Nodal
Server Edition	150	Valid	Never	PLDS Nodal
UMS Web Services	1000	Valid	Never	PLDS Nodal
Avaya Mac Softphone	1000	Valid	Never	PLDS Nodal
Avaya Softphone Licence	1000	Valid	Never	PLDS Nodal
SM Trunk Channels	128	Valid	Never	PLDS Nodal

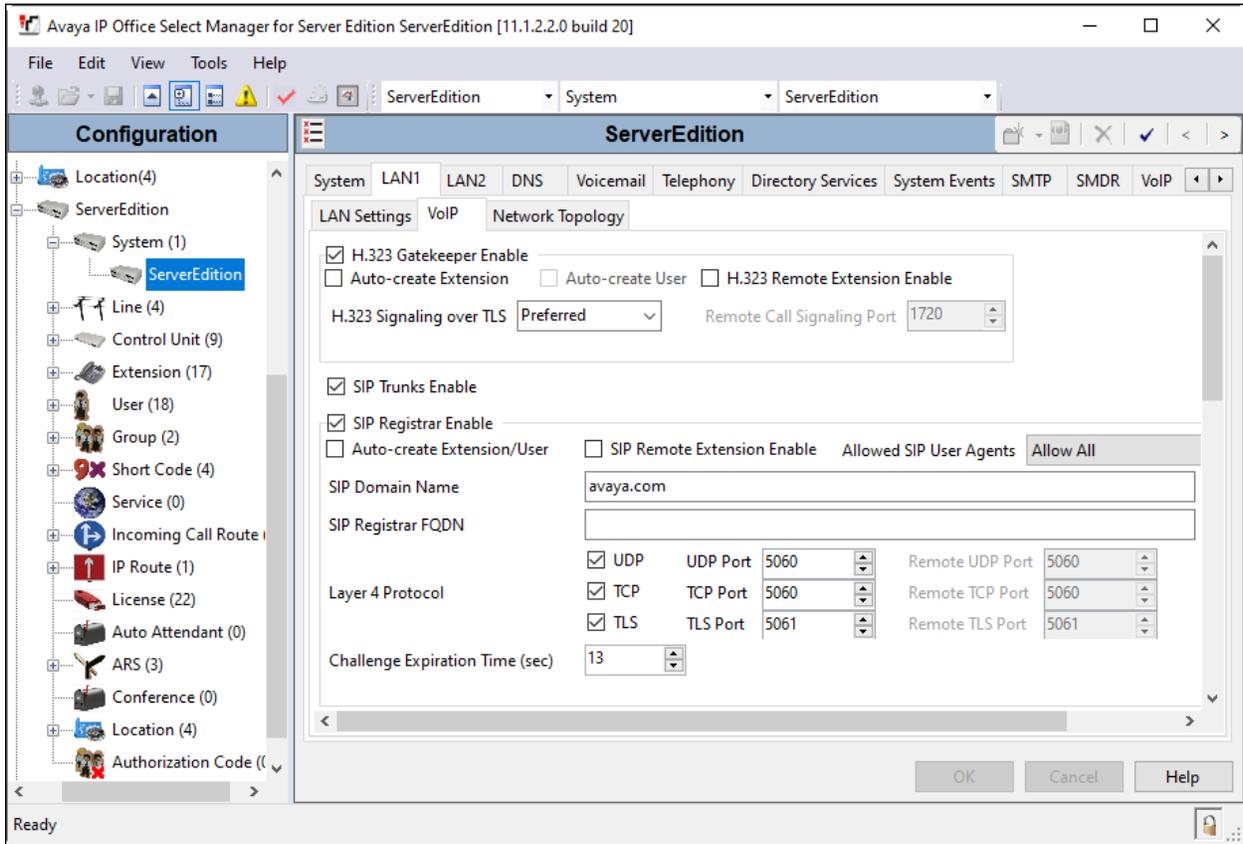
5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the System screen for the IP Office Server Edition in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the IP Address (e.g., *10.64.110.65*), which will be used in **Section 6.4** to configure MaxPlus Dispatch Console.



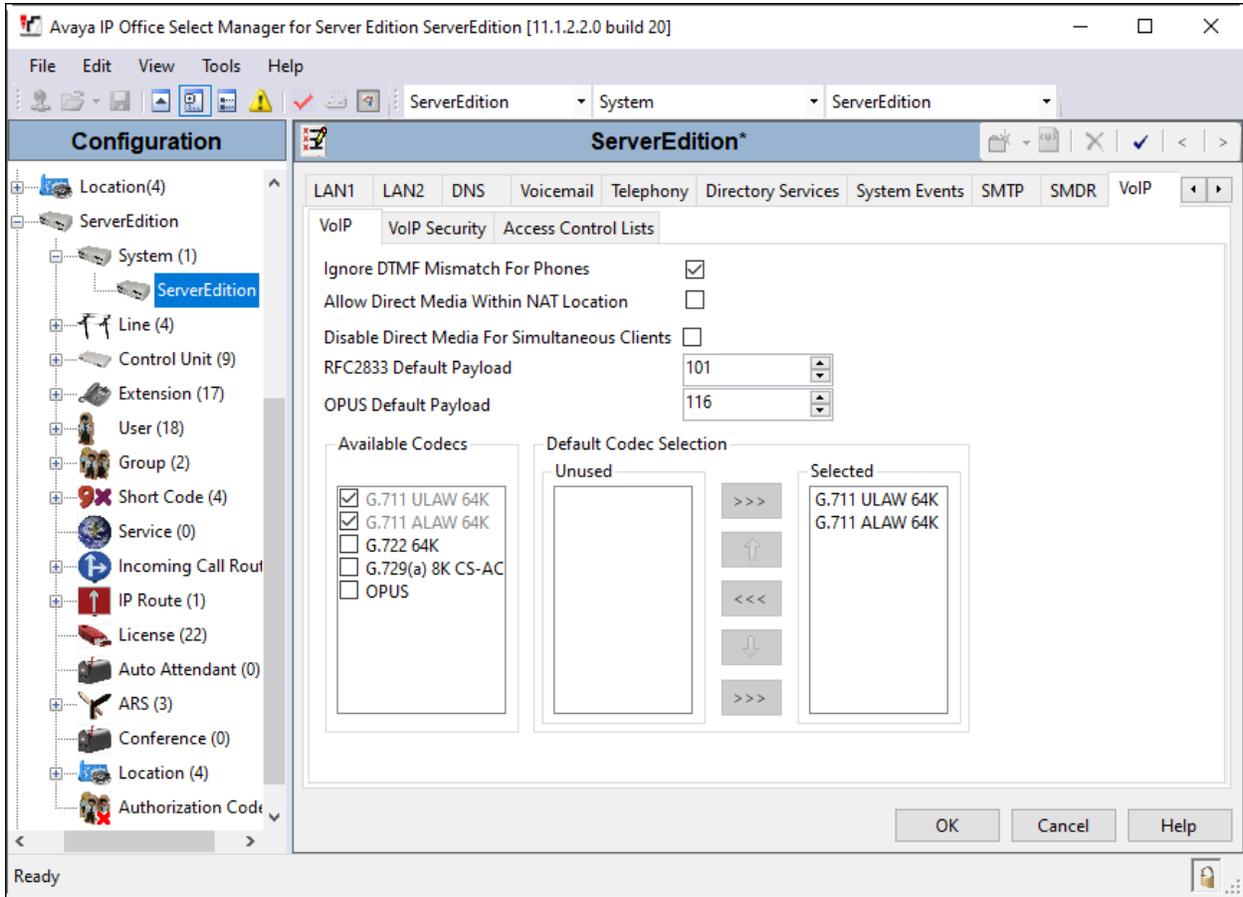
5.3. Administer SIP Registrar

Select the **VoIP** sub-tab in the **LAN1** tab. Ensure that **SIP Registrar Enable** is checked and enter a valid **Domain Name**. In the compliance testing, the **SIP Domain Name** field was set to *avaya.com*. TCP transport protocol was enabled for the **Layer 4 Protocol**, which was also used by MaxPlus Dispatch Console.



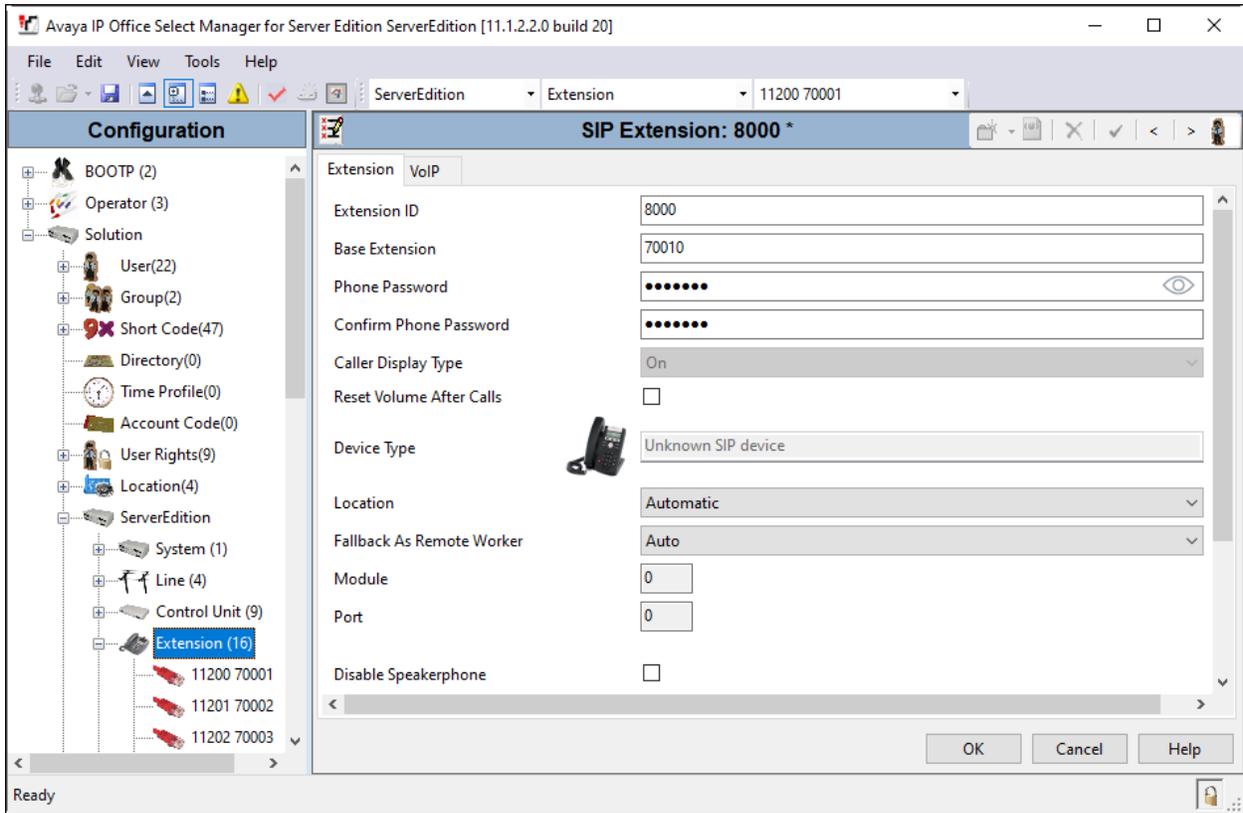
5.4. Administer IP Codecs

Select the **VoIP** tab in System. The **VoIP** sub-tab displays **Selected** codecs at the system level. *G.711 ULAW* and *G.711 ALAW* are selected. *G.729* can also be configured at the system level.

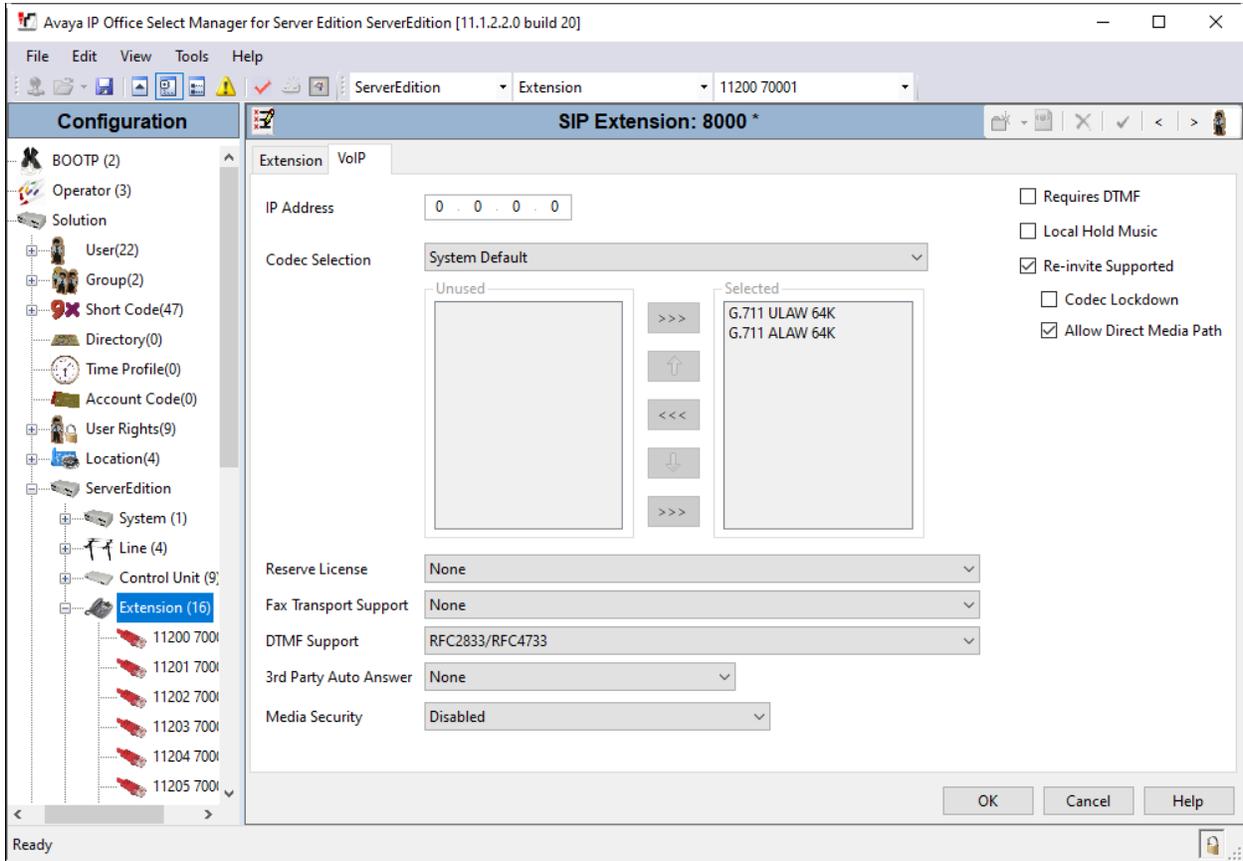


5.5. Administer SIP Extension for MaxPlus Dispatch Console

From the configuration tree in the left pane, right-click on **Extension** and select **New** → **SIP** from the pop-up list (not shown) to add a new SIP extension. Enter the desired extension for the **Base Extension** field as shown below. In this example, MaxPlus Dispatch Console was assigned extension **70010**. This is the extension that MaxPlus Dispatch Console will use to register with IP Office Server Edition. Enter an appropriate **Phone Password**. This will be used by MaxPlus Dispatch Console to register to IP Office Server.

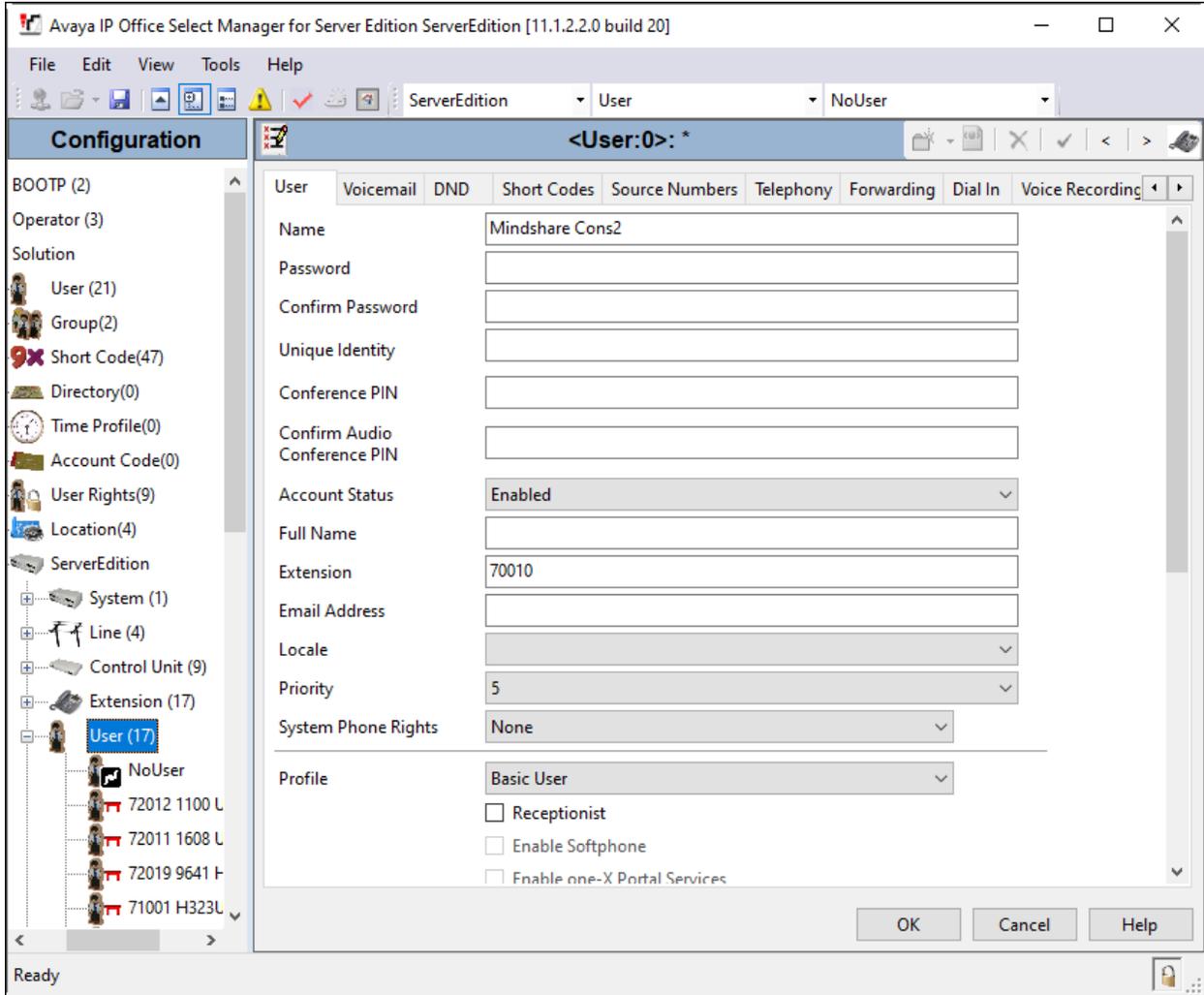


Select the **VoIP** tab. The **Codec Selection** is configured using the system level defaults from **Section 5.4** of *G.711 ULAW* and *G.711 ALAW*. Enable **Allow Direct Media Path** so that audio/RTP may flow directly between two SIP endpoints without using media resources in Avaya IP Office Server Edition. Select *disabled* for **Media Security**.

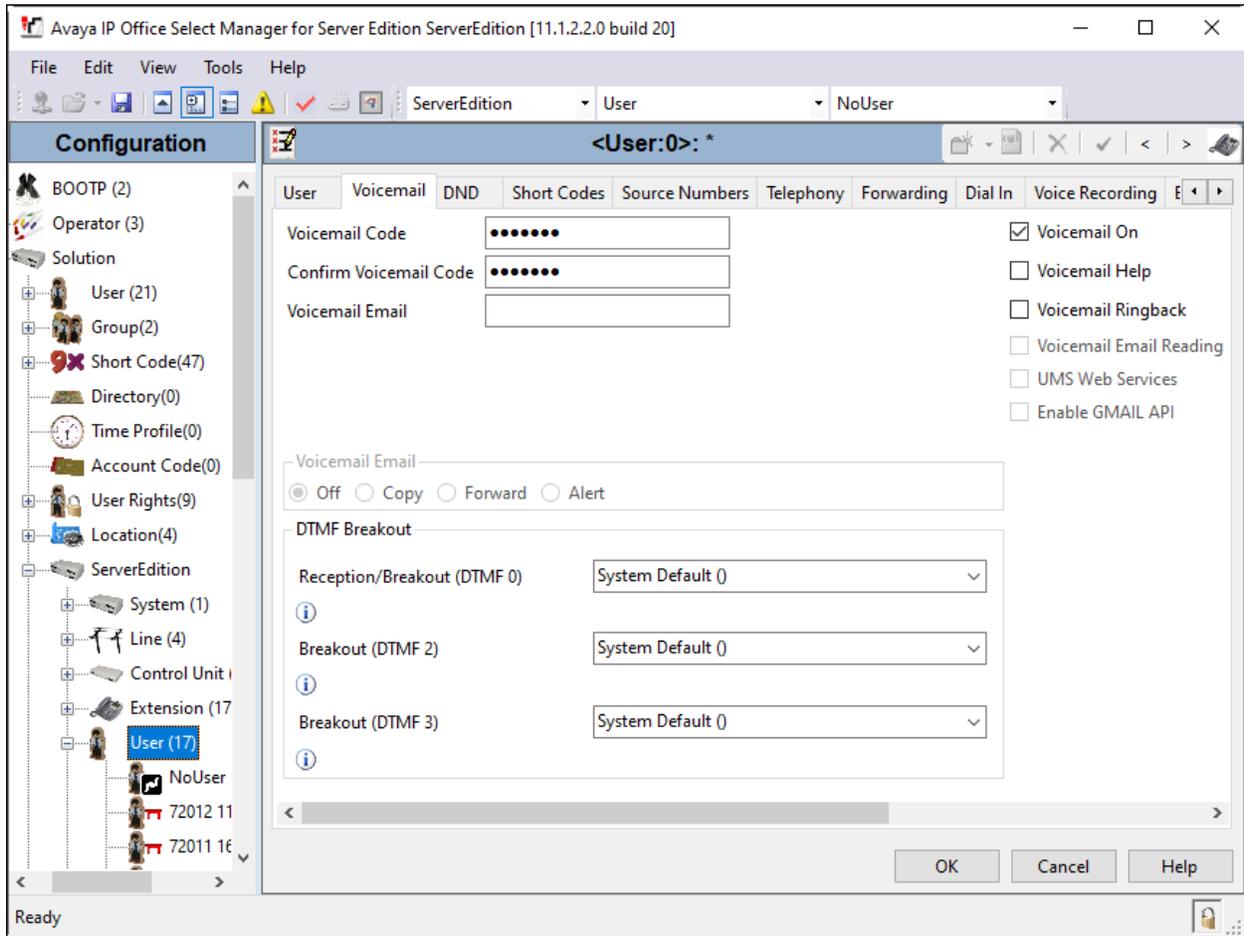


5.6. Administer SIP User for MaxPlus Dispatch Console

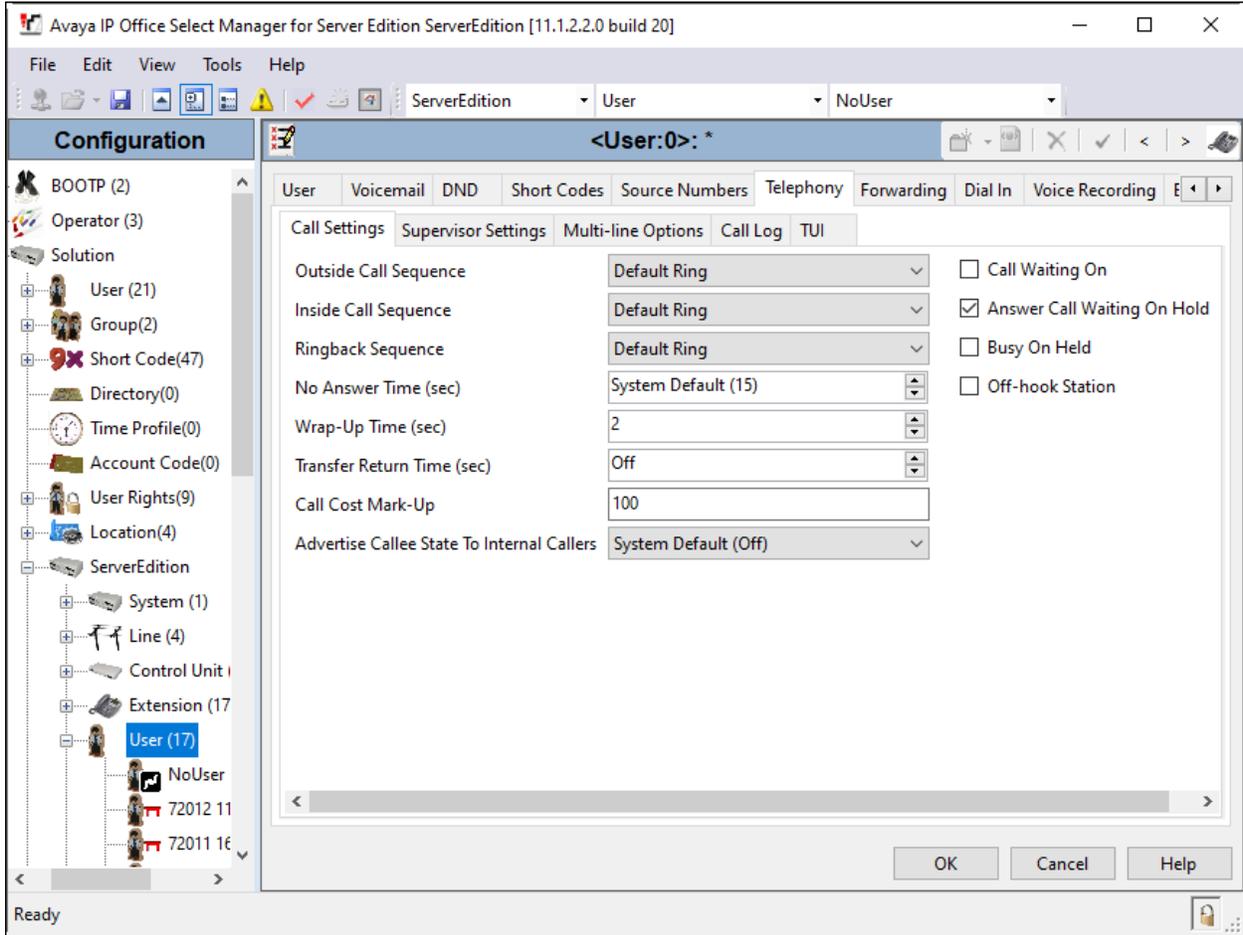
From the configuration tree in the left pane, right-click on **User** and select **New** from the pop-up list (not shown). Enter a value for the **Name** field (e.g., *Mindshare Cons2*). For the **Extension** field, enter the SIP extension administered in **Section 5.5** (e.g., *70010*).



Select the **Voicemail** tab and select **Voicemail On** to enable voicemail for MaxPlus Dispatch Console. Specify a **Voicemail Code** to be used when logging into voicemail.



Select the **Telephony** tab followed by the **Call Settings** sub-tab. Note the settings for the user.



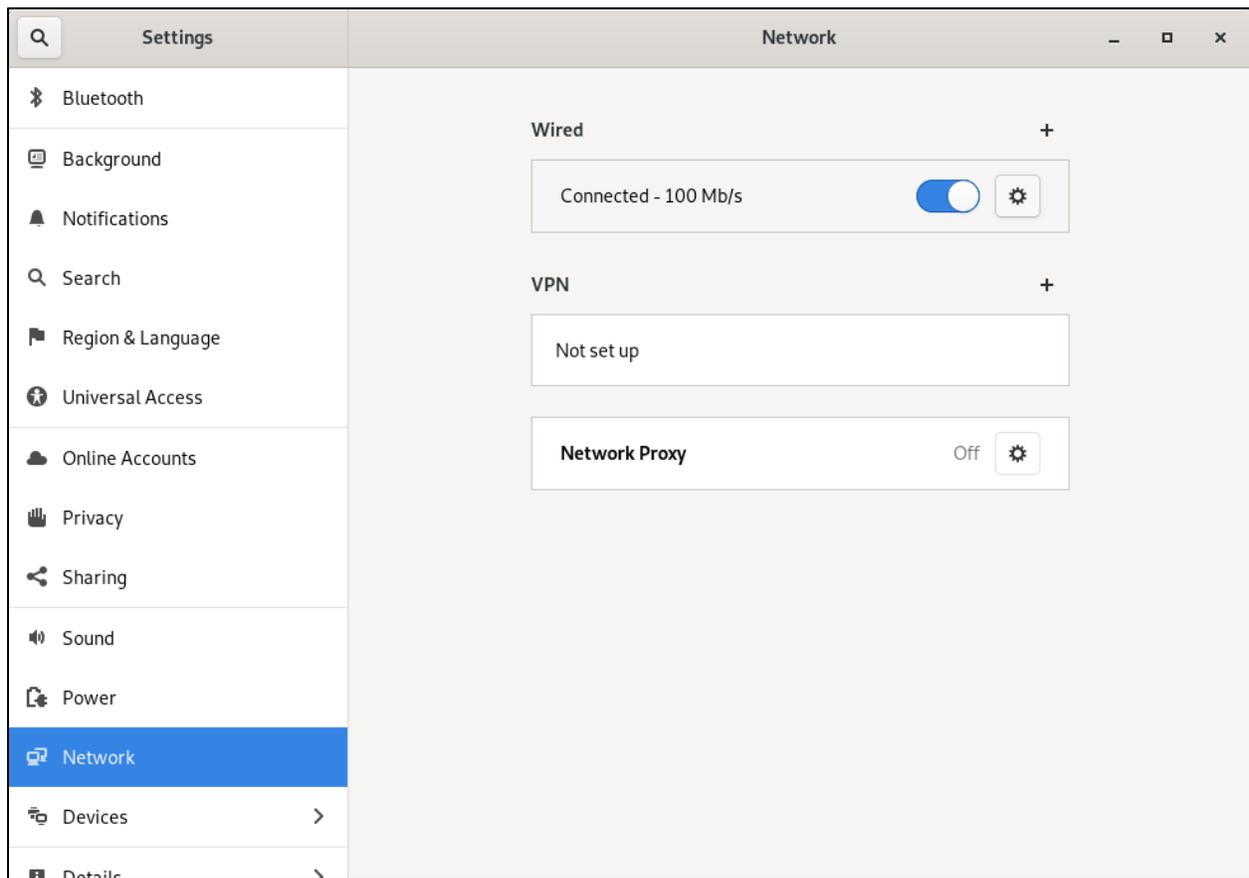
6. Configure CSS Mindshare 100500 MaxPlus Dispatch Console

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

6.1. Configure IP Address

Note: MaxPlus Dispatch Console requires two IP addresses. The second IP address is internally assigned as the next numerical assignment, e.g., if a static address of *10.64.10.51* is assigned to the console, *10.64.10.52* is internally assigned. Static IP addresses can avoid address conflicts. 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 to access the **Wired** properties.



Click the **Wired** Settings button to the right of the **Connected** slider. After the **Wired** settings appear, click on **IPv4**. Assign an address by clicking the **Manual** radio button and input the appropriate network information. In this case, an internal IP address of *10.64.10.51* is assigned (and implicitly *10.64.10.52*).

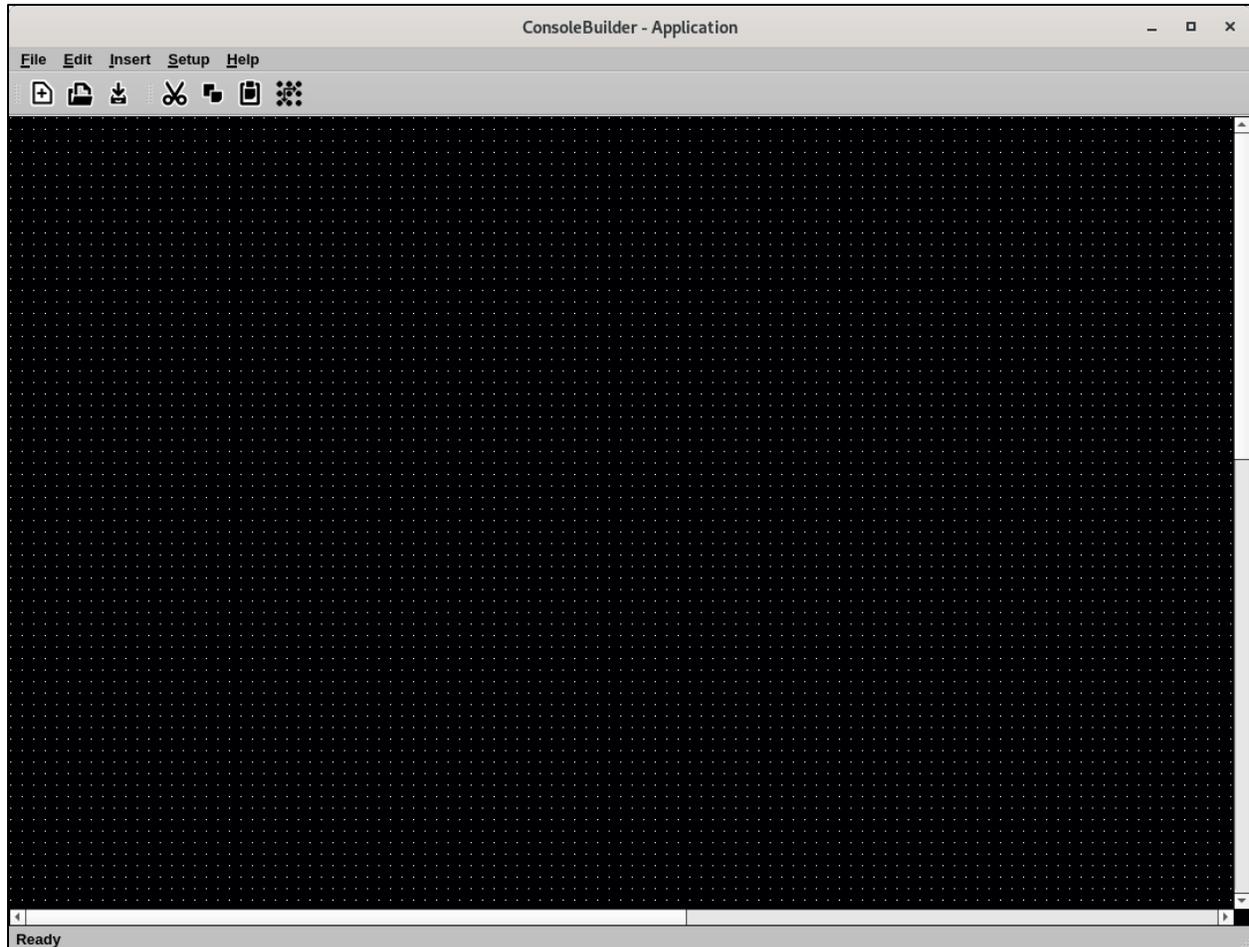
The screenshot shows the 'Wired' network configuration window. At the top, there are 'Cancel' and 'Apply' buttons. Below the title bar, there are tabs for 'Details', 'Identity', 'IPv4', 'IPv6', and 'Security'. The 'IPv4' tab is selected. Under 'IPv4 Method', the 'Manual' radio button is selected. The 'Addresses' section contains a table with columns for 'Address', 'Netmask', and 'Gateway'. The first row has the values '10.64.10.51', '255.255.255.0', and '10.64.10.1'. Below this, there is a 'DNS' section with a toggle set to 'Automatic' and a text input field containing '10.64.110.100, 75.75.75.75'. A note below the DNS field says 'Separate IP addresses with commas'. At the bottom, there is a 'Routes' section with a toggle set to 'Automatic' and an empty table with columns for 'Address', 'Netmask', 'Gateway', and 'Metric'.

Address	Netmask	Gateway
10.64.10.51	255.255.255.0	10.64.10.1

Address	Netmask	Gateway	Metric

6.2. 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.



6.3. Configure Phone System Parameters

Select **Setup** → **Setup Phone System** from the menu. Input 20 hops for **Set SIP Packet Time to live**. Select the **TCP** checkbox. Set **SIP Time Before Retry** to 2000 ms. Clear any default digit mappings. 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="2000"/>	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
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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:

6.4. Administer IP Comms

Select **Setup** → **Setup IP Comms** from the menu to administer lines on the console. Select *Phone* in the **Type** column. Input *70010* in the **Line Name** column. Select *uLaw* or *G.729* for the **Codec** column.

IP Address Setup Dialog ✕

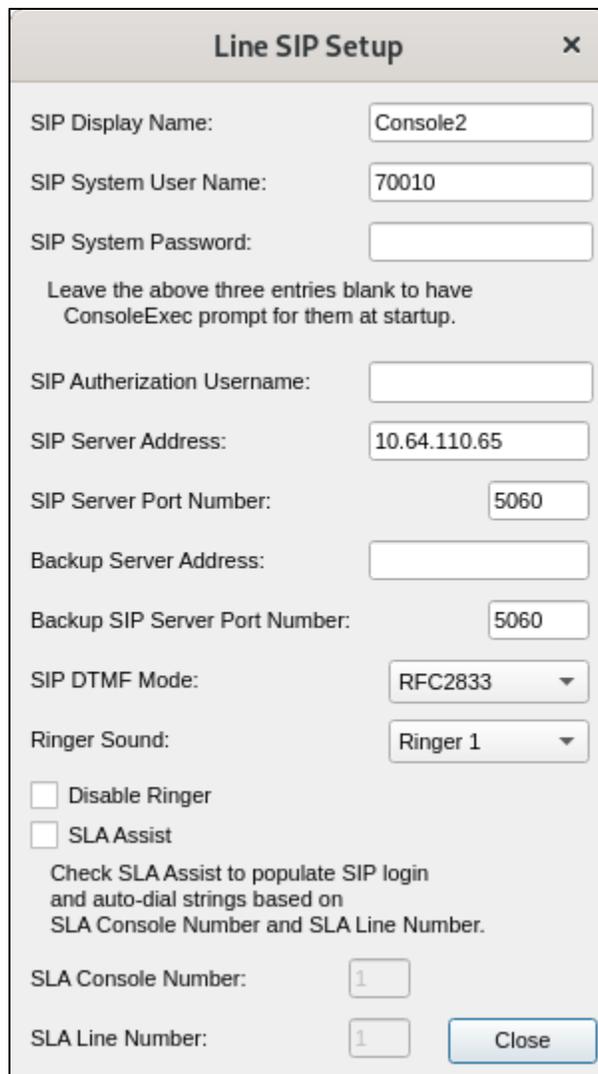
	Type	Line Name	RX IP Address	RX Port	TX IP Address	TX Port	Delay	Max Buffer Size	TTL	TxMon	Codec	Channel Items	RX Block	Other Setup
1	Phone	70010	235.98.99.101	10001	235.98.99.101	12001	5	40	2	On	uLaw	Setup SIP	RX Block	Other
2	Off	70112	235.98.99.102	10002	235.98.99.102	12002	5	40	2	On	uLaw	None	RX 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., *Console2*.
- **SIP System User Name:** Enter the extension from **Section 5.5** e.g., *70010*.
- **SIP System Password:** Enter the password for the extension administered in **Section 5.5**.
- **SIP Server Address:** Enter IP Office Server Edition IP address from **Section 5.2**, e.g., *10.64.110.65*.
- **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:

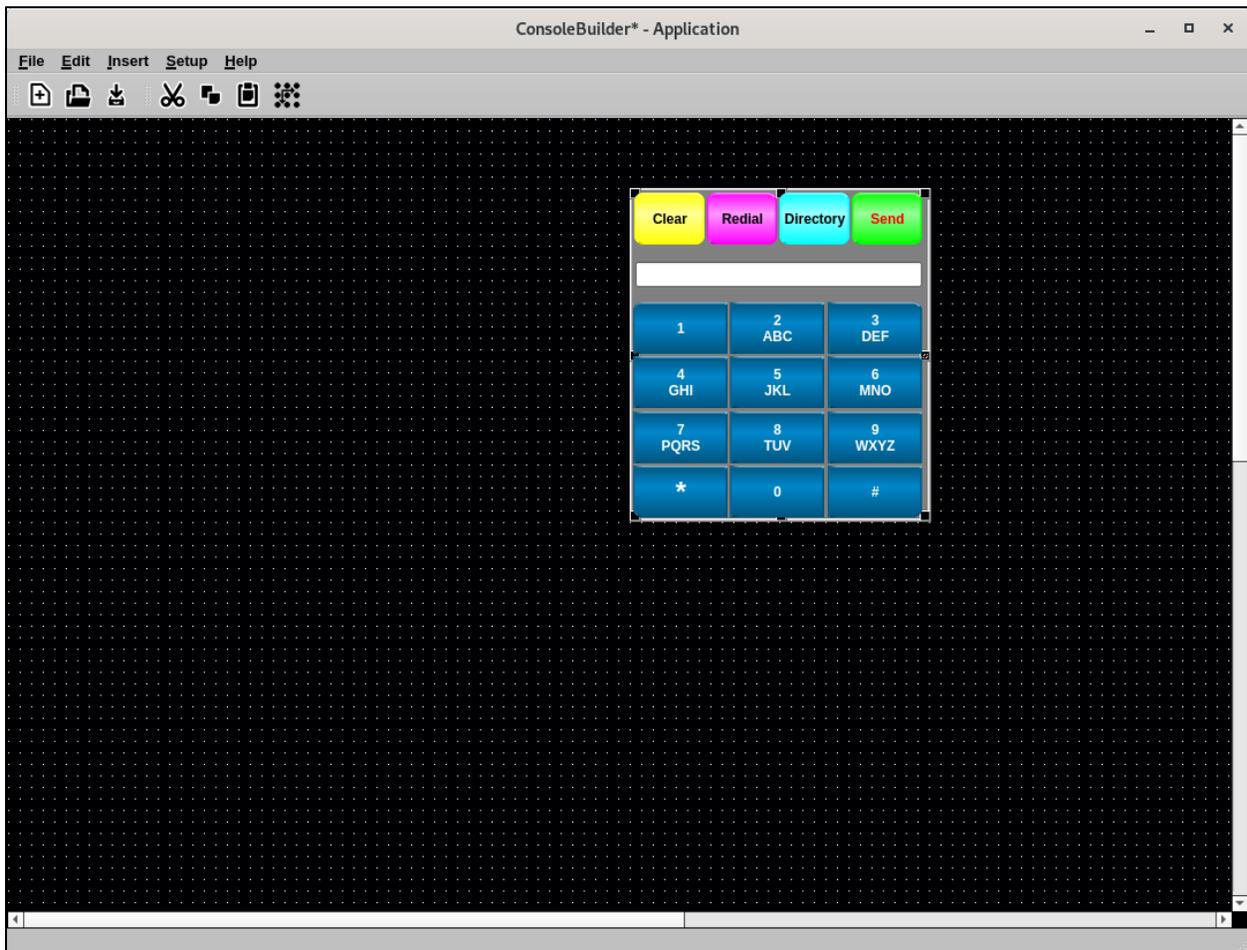
- SIP Display Name: Console2
- SIP System User Name: 70010
- SIP System Password: (empty)
- Leave the above three entries blank to have ConsoleExec prompt for them at startup.
- SIP Authorization Username: (empty)
- SIP Server Address: 10.64.110.65
- SIP Server Port Number: 5060
- Backup Server Address: (empty)
- Backup SIP Server Port Number: 5060
- SIP DTMF Mode: RFC2833
- Ringer Sound: Ringer 1
- Disable Ringer
- 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 button

6.5. Create Console Layout

Insert and configure console elements used for the VoIP user. Elements include the Dial Keypad, Line Indicator, Phone Line, Close Console, and Feature buttons. Feature buttons employed for interoperability testing included Mute, Hold, Transfer, Redial, Call Pickup Any, Call Park, and Call Unpark buttons. Console builder button controls provide a set of User Interface Functions that are assigned to buttons. These tools can be employed to provide additional functionality. Basic controls and a possible Call Park and Call Pickup button configuration are illustrated. For other button implementation configurations consult refer to [2].

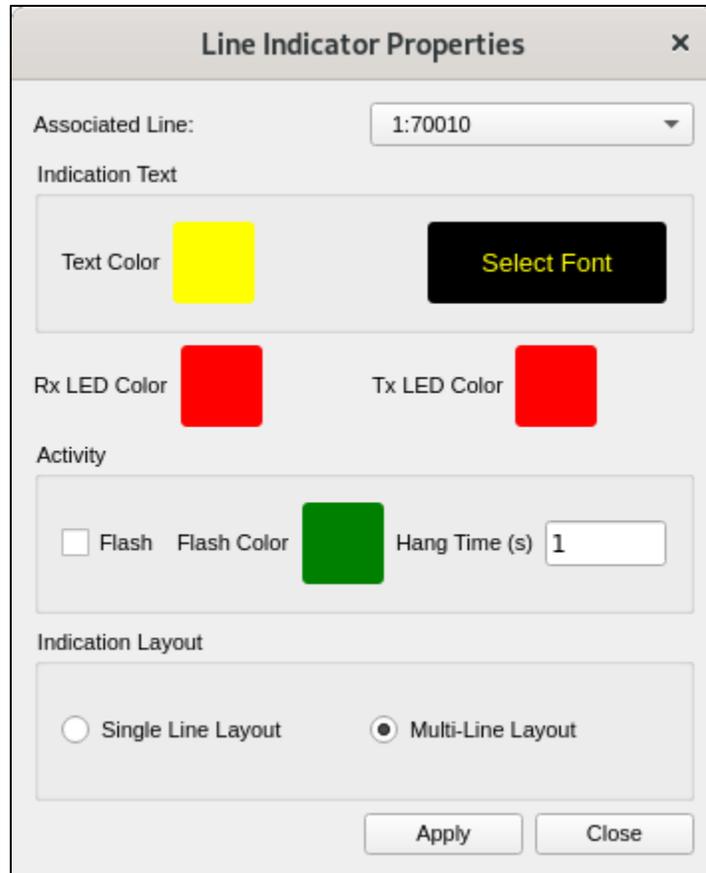
6.5.1. Dial Keypad

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



6.5.2. Line Indicator

Select **Insert** → **Insert Line Indicator**. Adjust the size and position of the Line Indicator on the grid. Mouse over the Line Indicator and right click on **Properties**. Select the Line defined in **Section 6.4** for **Associated Line**, e.g., *70010*.



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 "1:70010".
- Indication Text:** A section containing a "Text Color" selector with a yellow square, a "Select Font" button, 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 "Apply" and "Close" buttons.

6.5.3. Phone Line On/Offhook Button

Select **Insert** → **Insert Button Control**. Adjust the size and position of the On/Offhook button on the grid. Mouse over the new button and right click on **Properties**. Select the Line defined in **Section 6.4** for **Associated Line**, e.g., 70010. 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.

User Interface Button Property Setup Dialog

Primary Function

User Interface Function: Phone Line On/Offhook

Associated Line: 1:70010

Button ID: 0

Appearance Properties

Button Up Position

Button Color: [Dark Blue]

Text Color: [White]

Button Text: %VARLINENAME%
% OnHook

Icon Selection: [Phone Receiver Icon]

Button Down Position

Button Color: [Cyan]

Text Color: [Black]

Button Text: %VARLINENAME%
% OffHook

Icon Selection: [Phone Receiver Icon]

Corner Shapes

Square Upper Left Square Upper Right

Square Lower Left Square Lower Right

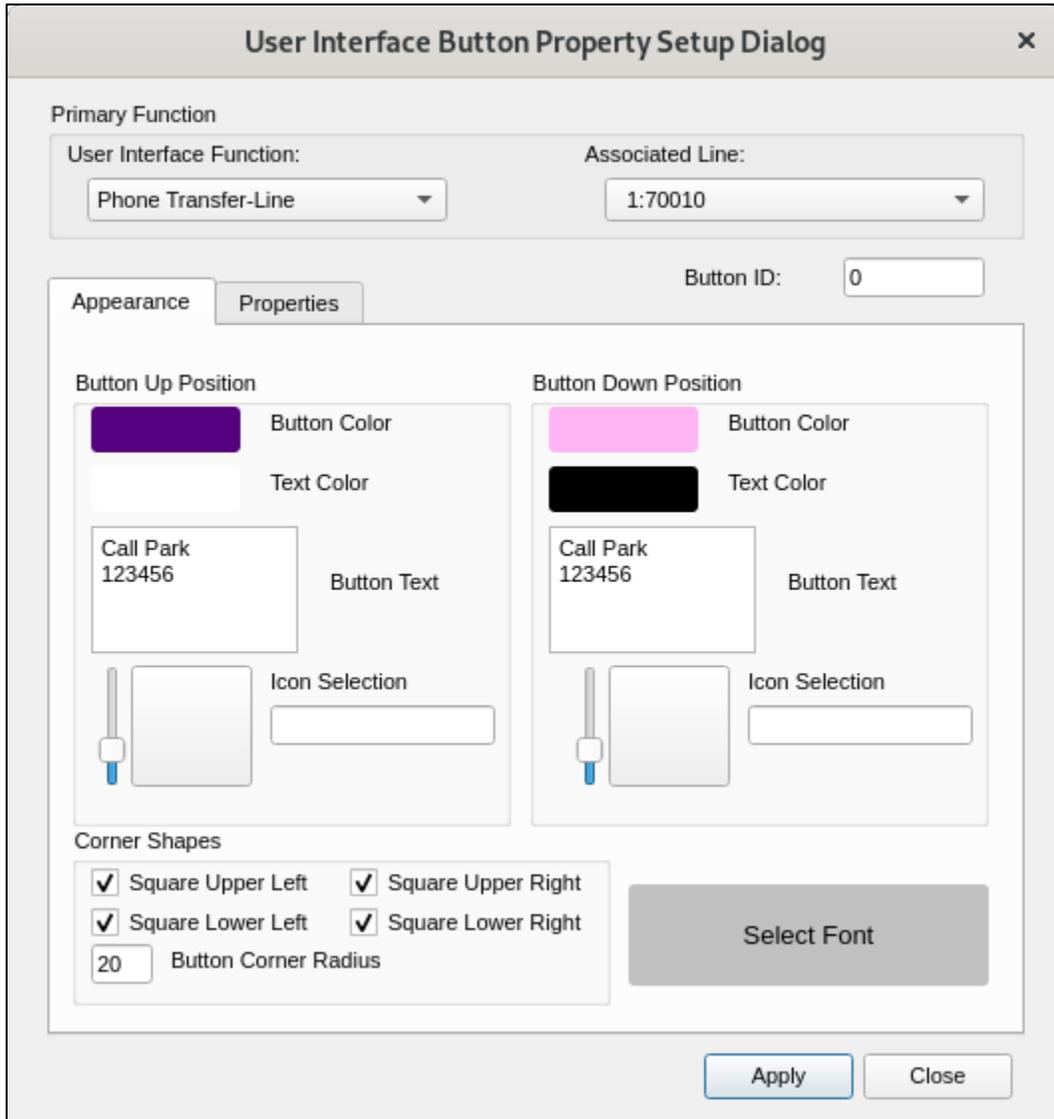
Button Corner Radius: 20

Select Font

Apply Close

6.5.4. Call Park Button

Select **Insert** → **Insert Button Control**. Adjust the size and position of the Call Park button on the grid. Mouse over the new button and right click on **Properties**. Select the Line defined in **Section 6.4** for **Associated Line**, e.g., *70010*. 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'. 'Associated Line' is set to '1:70010'.
- Button ID:** A text input field containing the number '0'.
- Appearance / Properties:** Two tabs are visible, with 'Properties' selected. This section is further divided into 'Button Up Position' and 'Button Down Position'.
 - Button Up Position:** Includes a 'Button Color' swatch (purple), a 'Text Color' swatch (white), a 'Button Text' input field containing 'Call Park 123456', and an 'Icon Selection' control with a slider and a selection box.
 - Button Down Position:** Includes a 'Button Color' swatch (pink), a 'Text Color' swatch (black), a 'Button Text' input field containing 'Call Park 123456', and an 'Icon Selection' control with a slider and a selection box.
- Corner Shapes:** Contains four checked checkboxes: 'Square Upper Left', 'Square Upper Right', 'Square Lower Left', and 'Square Lower Right'. Below them is a 'Button Corner Radius' input field set to '20'.
- Select Font:** A grey button labeled 'Select Font'.
- Buttons:** At the bottom right, there are 'Apply' and 'Close' buttons.

Select the **Properties** tab. Check **Enable Autodial**. Input the Short Code assigned to Call Park on IP Office, e.g., *37*123456#. In this case, the park slot number used is 123456.

User Interface Button Property Setup Dialog ✕

Primary Function

User Interface Function: Phone Transfer-Line ▾ Associated Line: 1:70010 ▾

Button ID: 0

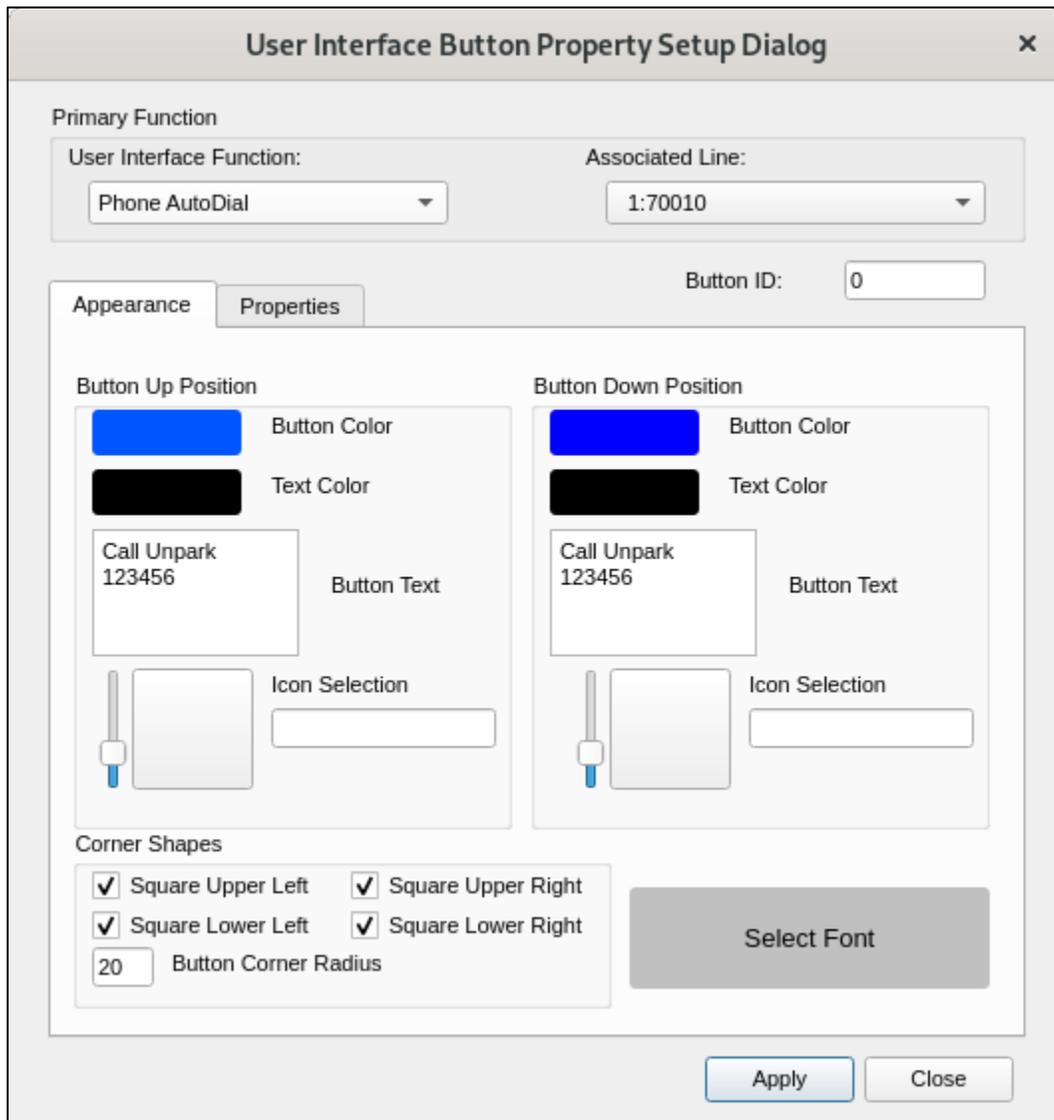
Appearance **Properties**

Property	Value(s)	Unit(s)
1	Enable Autodial: <input checked="" type="checkbox"/>	
2	Number: *37*123456#	
3	Transfer Blind: <input 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

6.5.5. Call Unpark Button

Select **Insert** → **Insert Button Control**. Adjust the size and position of the Call Unpark button on the grid. Mouse over the new button and click properties. Select the Line defined in **Section 6.4** for **Associated Line**, e.g., *70010*. 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 (X). Below the title bar, there are two dropdown menus: 'User Interface Function' set to 'Phone AutoDial' and 'Associated Line' set to '1:70010'. 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), and a 'Button Text' field containing 'Call Unpark 123456'. Below the text fields are 'Icon Selection' controls, each consisting of a vertical slider and a text input field. 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'. 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 very bottom of the dialog are 'Apply' and 'Close' buttons.

Select the **Properties** tab. Check **Dial on Associated Line**. Input the short Code to unpark the call on IP Office for **Dial String when Clicked**, e.g., **38*123456#* that uses the analogous park slot number *123456* used in **Section 6.5.4**.

User Interface Button Property Setup Dialog ✕

Primary Function

User Interface Function: Phone AutoDial Associated Line: 1:70010

Button ID: 0

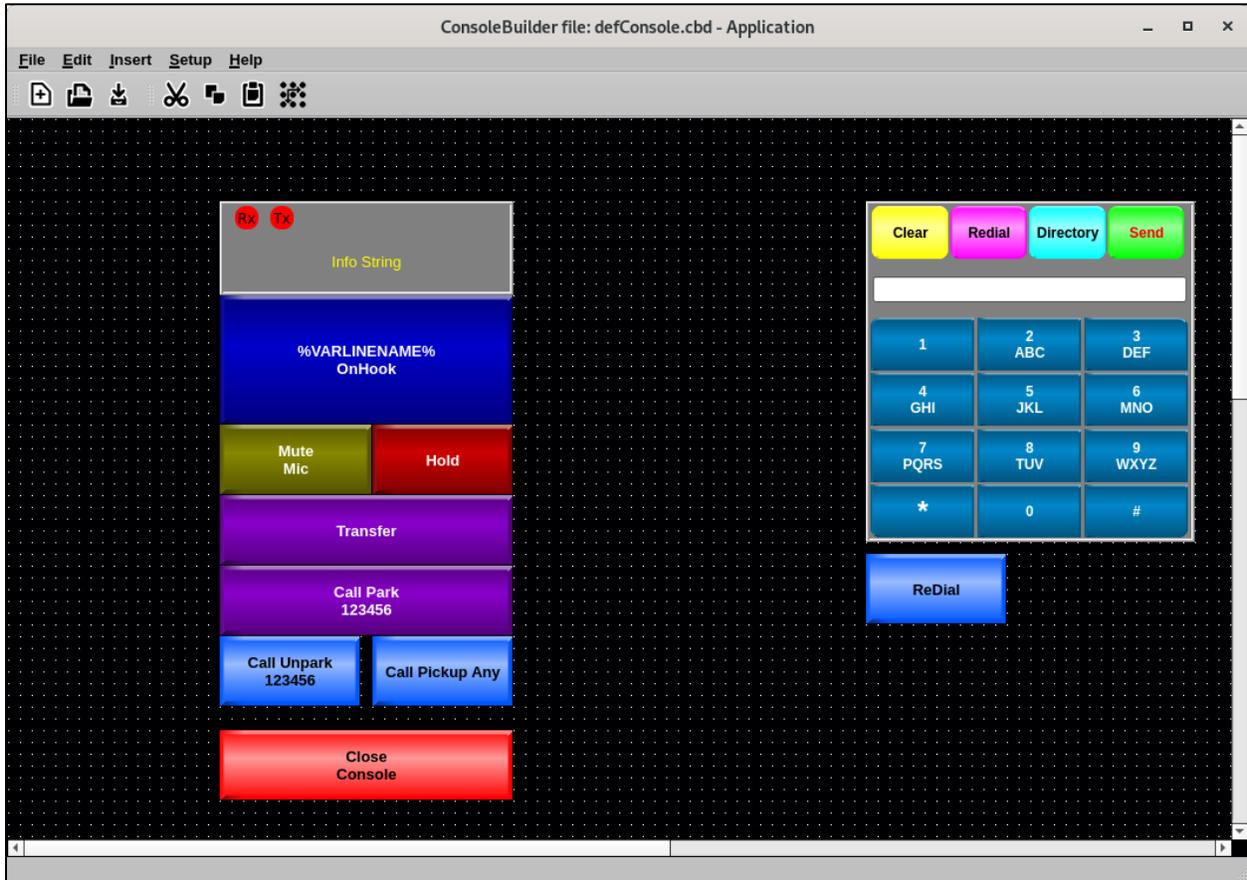
Appearance **Properties**

Property	Value(s)	Unit(s)
1	Dial String when Clicked: *38*123456#	
2	Preset Dial String: <input type="checkbox"/>	
3	Preset String: 	
4	Popup Dialpad on Click: <input type="checkbox"/>	
5	Dial on Associated Line: <input checked="" type="checkbox"/>	
6	Enable Autodial #2: <input type="checkbox"/>	
7	Enable Preset Dial #2: <input type="checkbox"/>	
8	Number 2: 	
9		
10		
11		
12		
13		
14		
15		
16		

Apply
Close

6.6. 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`.

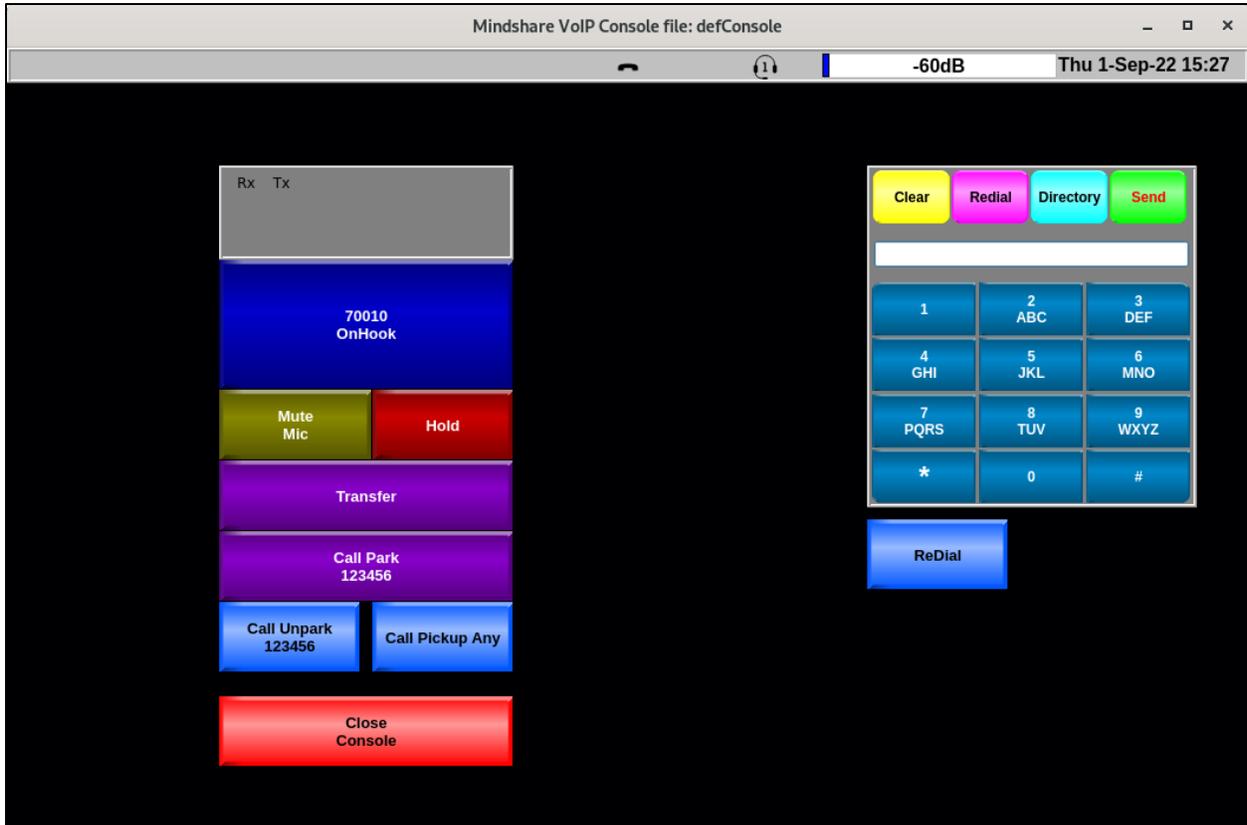


7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of MaxPlus Dispatch Console with IP Office.

7.1. Launch Console

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



7.2. Registration Status

Verify that MaxPlus Dispatch Console has successfully registered with IP Office. From a PC with **IP Office Admin Suite** installed, invoke **IP Office System Status**, navigate to the MaxPlus Dispatch Console SIP extension and verify **Media Stream** is set to *RTP*, **Layer 4 Protocol** is set to *TCP*, and **Current State** is shown as *Idle*.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - ServerEdition (10.64.110.65) - IP Office Linux PC 11.1.2.2.0 build 20". The main window has the Avaya logo and the title "IP Office System Status". A menu bar includes "Help", "Snapshot", "LogOff", "Exit", and "About".

The left sidebar contains a tree view with the following items: System, Alarms (3), Extensions (3) (with a sub-item 70010 selected), Trunks (4), Active Calls, Resources, Voicemail, and IP Networking Locations.

The main content area is titled "Extension Status" and displays the following configuration for extension 70010:

- Extension Number: 70010
- IP address: 10.64.10.51
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: ConsoleExec_v3.27.2-release_by_Mindshare
- Media Stream: RTP
- Layer 4 Protocol: TCP
- Current User Extension Number: 70010
- Current User Name: Mindshare Cons2
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Number of New Messages: 0
- Phone Manager Type: None
- SIP Device Features: REFER
- License Reserved: No
- Last Date and Time License Allocated: 9/6/2022 10:13:24 AM
- DTMF Required: No
- Packet Loss Fraction: (empty)
- Jitter: (empty)
- Round Trip Delay: (empty)
- Connection Type: (empty)
- Codec: (empty)
- Remote Media Address: (empty)

Below the configuration is a table with the following columns: Call Ref, Current State, Time in State, Calling Number or Called Number, Direction, and Other Party on Call. The table contains one row with the following data:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
	Idle	00:00:32			

At the bottom of the window, there are several buttons: Trace, Trace All, Pause, Ping, Call Details, Print..., and Save As... The status bar at the bottom right shows the time "1:13:29 PM" and the status "Online".

7.3. Basic Calls

Establish a call between MaxPlus Dispatch Console and a local Avaya SIP desk phone. In **IP Office System Status**, navigate to the MaxPlus Dispatch Console SIP extension and verify that the **Current State** is *Connected* as shown. Verify two-way audio.

The screenshot shows the Avaya IP Office System Status application window. The title bar reads "Avaya IP Office System Status - ServerEdition (10.64.110.65) - IP Office Linux PC 11.1.2.2.0 build 20". The main header displays the Avaya logo and "IP Office System Status". A menu bar includes "Help", "Snapshot", "LogOff", "Exit", and "About".

The left sidebar contains a navigation tree with the following items: System, Alarms (3), Extensions (3) (with 70010 selected), 72016, 72019, Trunks (4), Active Calls, Resources, Voicemail, IP Networking, and Locations.

The main content area is titled "Extension Status" and displays the following details for extension 70010:

- Extension Number: 70010
- IP address: 10.64.10.51
- Standard Location: None
- Registrar: Primary
- Telephone Type: Unknown SIP Device
- User-Agent SIP header: ConsoleExec_v3.27.2-release_by_Mindshare
- Media Stream: RTP
- Layer 4 Protocol: TCP
- Current User Extension Number: 70010
- Current User Name: Mindshare Cons2
- Forwarding: Off
- Twinning: Off
- Do Not Disturb: Off
- Message Waiting: Off
- Number of New Messages: 0
- Phone Manager Type: None
- SIP Device Features: REFER
- License Reserved: No
- Last Date and Time License Allocated: 9/6/2022 10:13:24 AM
- DTMF Required: No
- Packet Loss Fraction: 0%
- Jitter: 0ms
- Round Trip Delay: 0ms
- Connection Type: VCM (SRTP)
- Codec: G711 Mu
- Remote Media Address: 10.64.10.225

Below the extension details is a call log table:

Call Ref	Current State	Time in State	Calling Number or Called Number	Direction	Other Party on Call
610	Connected	00:00:25		Outgoing	Extn 72019, 9641 H323 User

At the bottom of the window, there are buttons for "Trace", "Trace All", "Pause", "Ping", "Call Details", "Print...", and "Save As...". The status bar at the bottom right shows the time "11:34:55 AM" and the status "Online".

8. Conclusion

These Application Notes have described the administration steps required to integrate CSS Mindshare MaxPlus 100500 Dispatch Console 3.27.2 with Avaya IP Office Server Edition 11.1 and Avaya IP Office 500 V2 Expansion System 11.1. CSS Mindshare 100500 MaxPlus Dispatch Console successfully registered with IP Office as a SIP user, and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

9. Additional References

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

[1] *Administering Avaya IP Office Platform™ with Manager*

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

[2] *MS0101_UM_ConsoleApplicationManual*, Revision 1.15, June 23, 2022

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