Avaya Solution & Interoperability Test Lab

Application Notes for Extron DMP 128 Plus C V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Extron DMP 128 Plus C V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Extron DMP 128 Plus C V is a digital matrix processor suitable for conferencing applications. These Application Notes also apply to the Extron DMP 128 Plus C V AT (although not explicitly tested), which only differs in that it provides DANTE support.

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

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

1. Introduction

These Application Notes describe the configuration steps required to integrate Extron DMP 128 Plus C V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Extron DMP 128 Plus C V is a digital matrix processor suitable for conferencing applications. Configuration is performed via the Extron DSP Configurator, which provides a GUI for easy visualization of all the signal paths within a single window and the ability to adjust all input levels, DSP processing parameters, mixing points, and output levels. Extron DMP 128 Plus C V registers to Avaya Aura® Session Manager as a SIP endpoint. These Application Notes also apply to the Extron DMP 128 Plus C V AT (although not explicitly tested), which only differs in that it provides DANTE support.

The Extron DMP Plus Series also includes the products detailed in **Attachment 1**. Since the products share the same firmware version, these Application Notes also apply to them.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between DMP 128 Plus, Avaya SIP/H.323 desk phones and the PSTN, and exercising basic telephony features, such as hold, mute, and conference. Additional telephony features, such as call forward, follow me, and call pickup were also verified using Communication Manager Features Access Codes (FACs).

The serviceability testing focused on verifying that DMP 128 Plus returned to service after re-connecting the Ethernet cable or rebooting DMP 128 Plus.

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

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

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

For the testing associated with these Application Notes, the interface between Avaya systems and Extron DMP 128 Plus C V did not include use of any specific encryption features as requested by Extron.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of DMP 128 Plus with Session Manager.
- Calls between DMP 128 Plus and Avaya SIP/H.323 deskphones with Direct IP Media (Shuffling) disabled.
- Calls between DMP 128 Plus and the PSTN.
- UDP transport protocol.
- Support of G.711μ-law, G.729, and G.722 codecs.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, conference, and long duration calls.
- Extended telephony features using Communication Manager FACs for Call Forward, Follow Me, Call Unpark, and Call Pickup.
- Proper system recovery after a restart of DMP 128 Plus and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- During the compliance test, the Phone Dialer tool, available through the Extron DSP Configurator, was used for placing and answering calls. The Phone Dialer is designed for basic test purposes only. Typically, customers would use the Extron CCI Pro 700 TouchLink Pro Conference Room Control Interface, or one of Extron's many other customizable touchpanel devices, which provides a more robust experience and audio tone feedback for each call.
- For this solution, Direct IP Media (Shuffling) should be disabled for calls to the DMP 128 Plus. Currently, DMP 128 Plus doesn't support receiving a re-INVITE without SDP, which could have adverse effects on shuffled calls and various hold scenarios.
- DMP 128 Plus does not support call transfer.
- DMP 128 Plus supported conferencing by configuring the DSP to automatically mix audio from all active call appearances.
- Only one codec should be configured on DMP 128 Plus for compatibility with Communication Manager to prevent audio issues during call establishment.

2.3. Support

For technical support on the Extron DMP 128 Plus C V, contact the Extron Support Hotline via phone or website.

- **Phone:** +1 (800) 633-9876
- **Web:** https://www.extron.com/company/contactus.aspx

3. Reference Configuration

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

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway. The Avaya G450 Media Gateway was connected to the PSTN via an ISDN-PRI trunk (not shown).
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J100 Series SIP Deskphones.
- Extron DMP 128 Plus C V, Extron DSP Configurator with Phone Dialer for establishing calls.

Extron DMP 128 Plus C V registered with Session Manager and was configured as Off-PBX Stations (OPS) on Communication Manager.

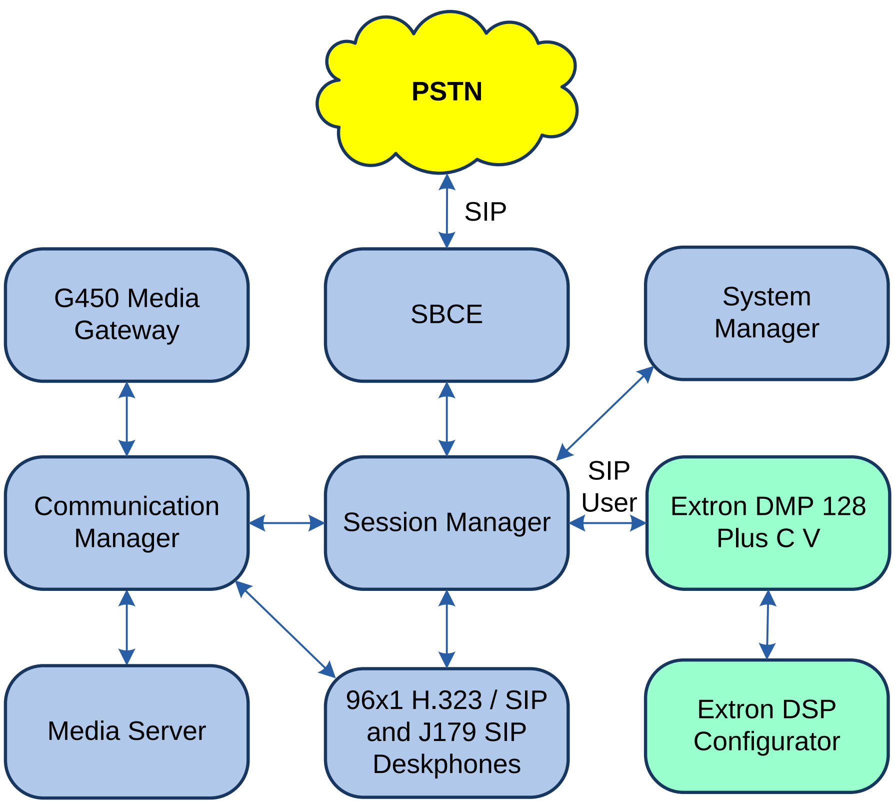


Figure 1: Avaya SIP-based Network with Extron DMP 128 Plus C V

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	8.1.1.0.0-FP1
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.1.121
Avaya Aura® Session Manager	8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.079814
Avaya Aura® System Manager	8.1.0.0.810007
Avaya 96x1 Series IP Deskphone	6.8304 (H.323) 7.1.7.0.11 (SIP)
Avaya J100 Series SIP Deskphone	4.0.3.1.4
Extron DMP 128 Plus C V	v1.06.0001-b001
Extron DSP Configurator	2.23.1.42

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 and 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 Extron DMP 128 Plus C V is configured through System Manager in **Section 6.2**.

5.1. Verify License

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

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

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

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

                                                                USED
                              Platform Maximum Ports:  48000     86
                                    Maximum Stations:  36000     25
                            Maximum XMOBILE Stations:  36000      0
                  Maximum Off-PBX Telephones - EC500:  41000      0
                  Maximum Off-PBX Telephones -   OPS:  41000     14
                  Maximum Off-PBX Telephones - PBFMC:  41000      0
                  Maximum Off-PBX Telephones - PVFMC:  41000      0
                  Maximum Off-PBX Telephones - SCCAN:      0      0
                       Maximum Survivable Processors:  313        0

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

5.2. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. **IP-IP Direct Audio** (shuffling) should be disabled to enforce media resources in the Avaya G450 Media Gateway or Avaya Aura® Media Server to be used. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager.

Note: Refer to the second bullet item in **Section 2.2** on shuffling.

```
change ip-network-region 1                                          Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: avaya.com
    Name:                       Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 50999
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to DMP 128 Plus. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. DMP 128 Plus was tested using G.711MU, G.729, and G.722 codecs. The following IP Codec Set is shown configured with the G.711MU codec.

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

                          IP MEDIA PARAMETERS
    Codec Set: 1

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

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

6. Configure Avaya Aura® Session Manager

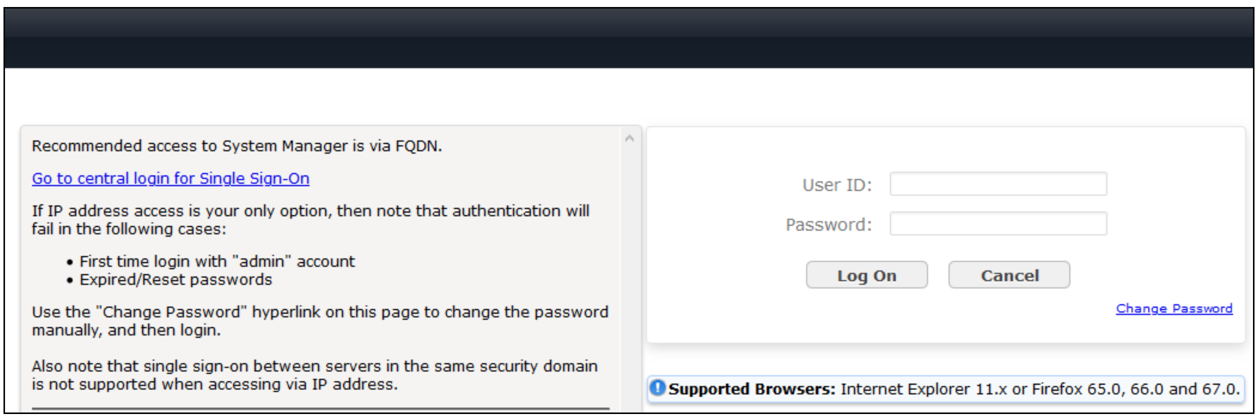
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for DMP 128 Plus
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for DMP 128 Plus.

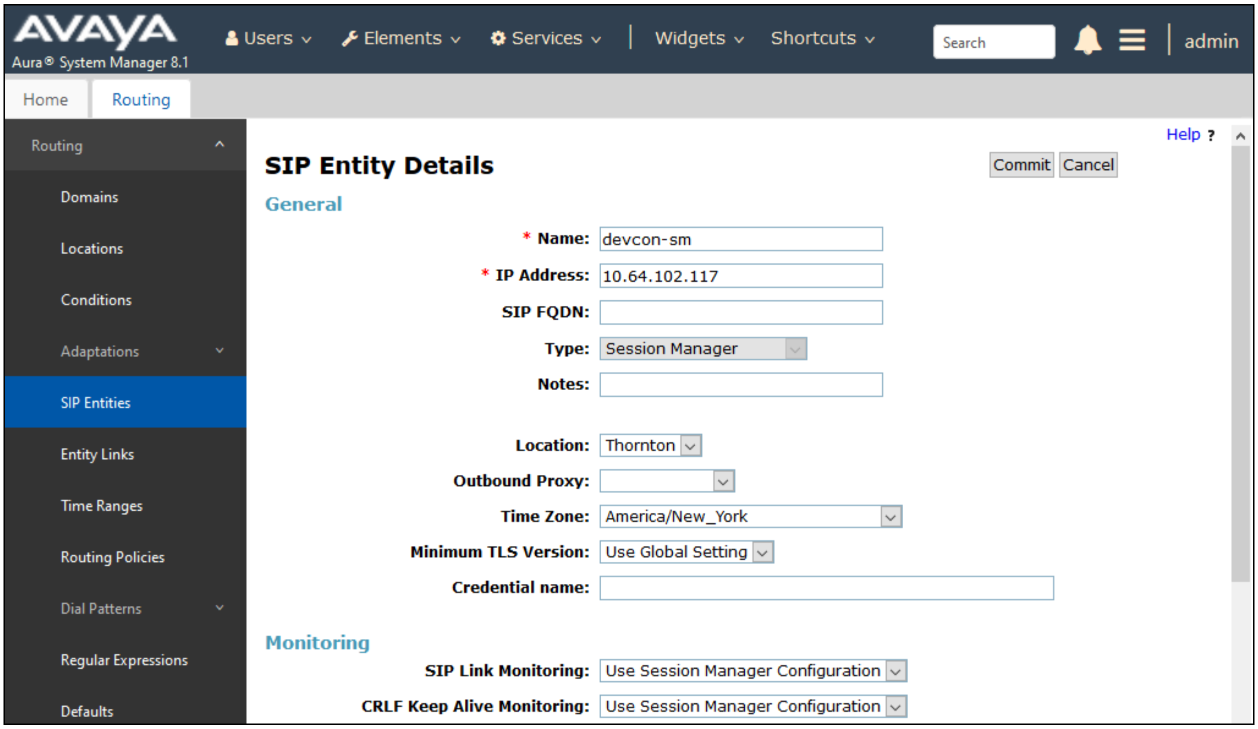
6.1. Launch System Manager

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

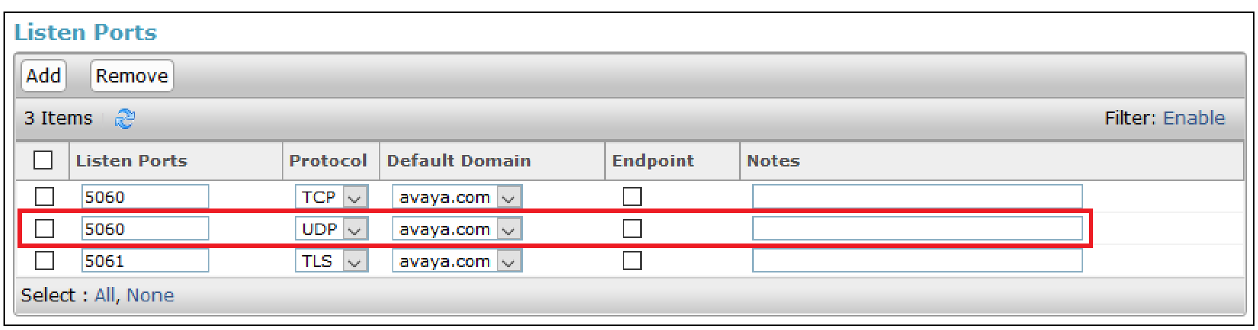


6.2. Set Network Transport Protocol for Extron DMP 128 Plus

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

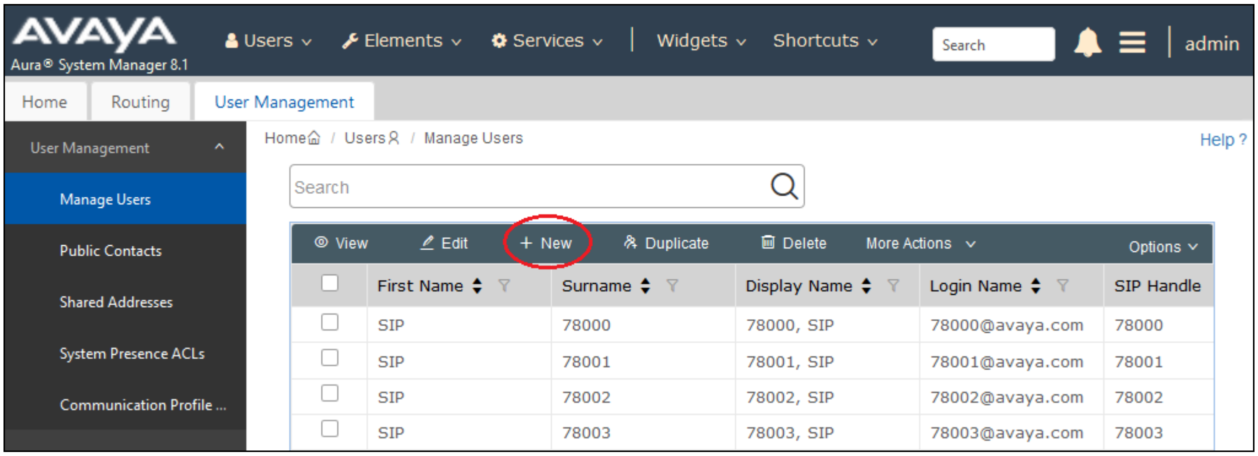


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by DMP 128 Plus is specified in the list below. For the compliance test, DMP 128 Plus used UDP network transport.



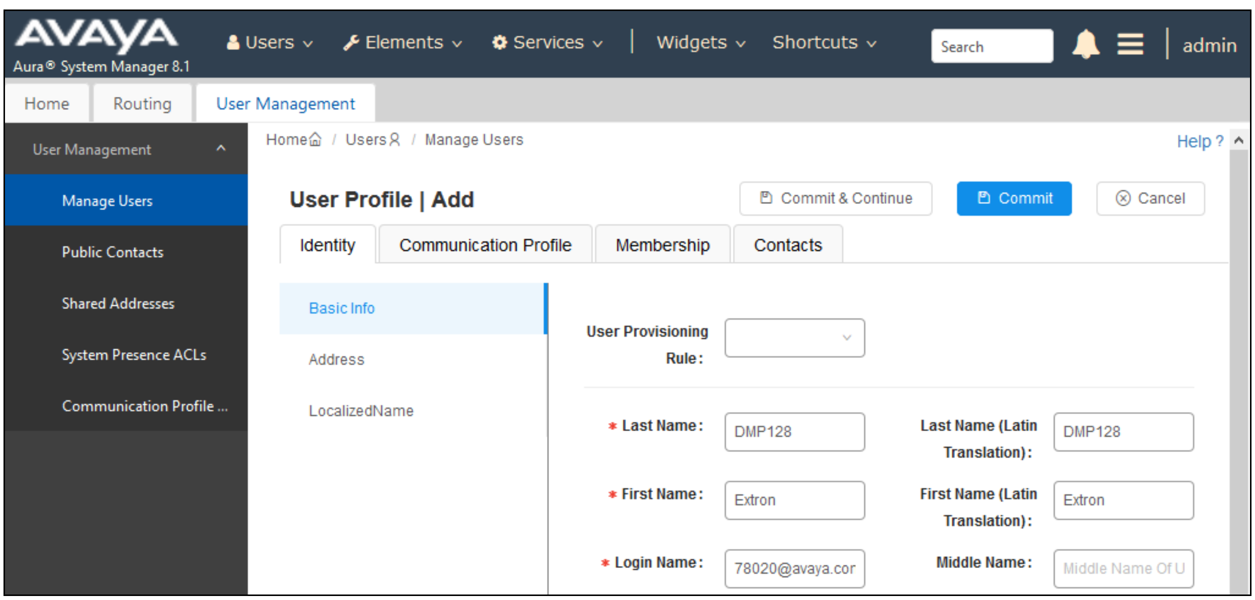
6.3. Administer SIP User

In the subsequent screen (not shown), select **Users → User Management → Manage Users** to display the **User Management** screen below. Click **New** to add a user.



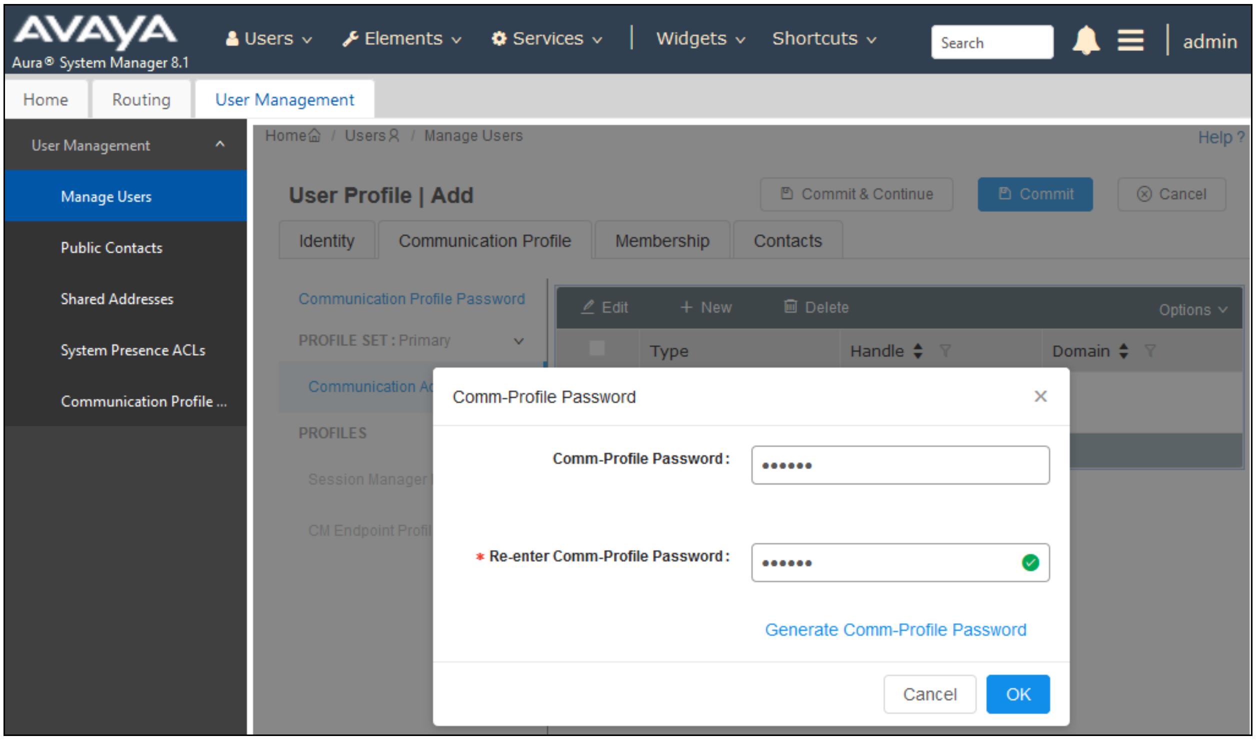
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "*<ext>@<domain>*", where "*<ext>*" is the desired DMP 128 Plus SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



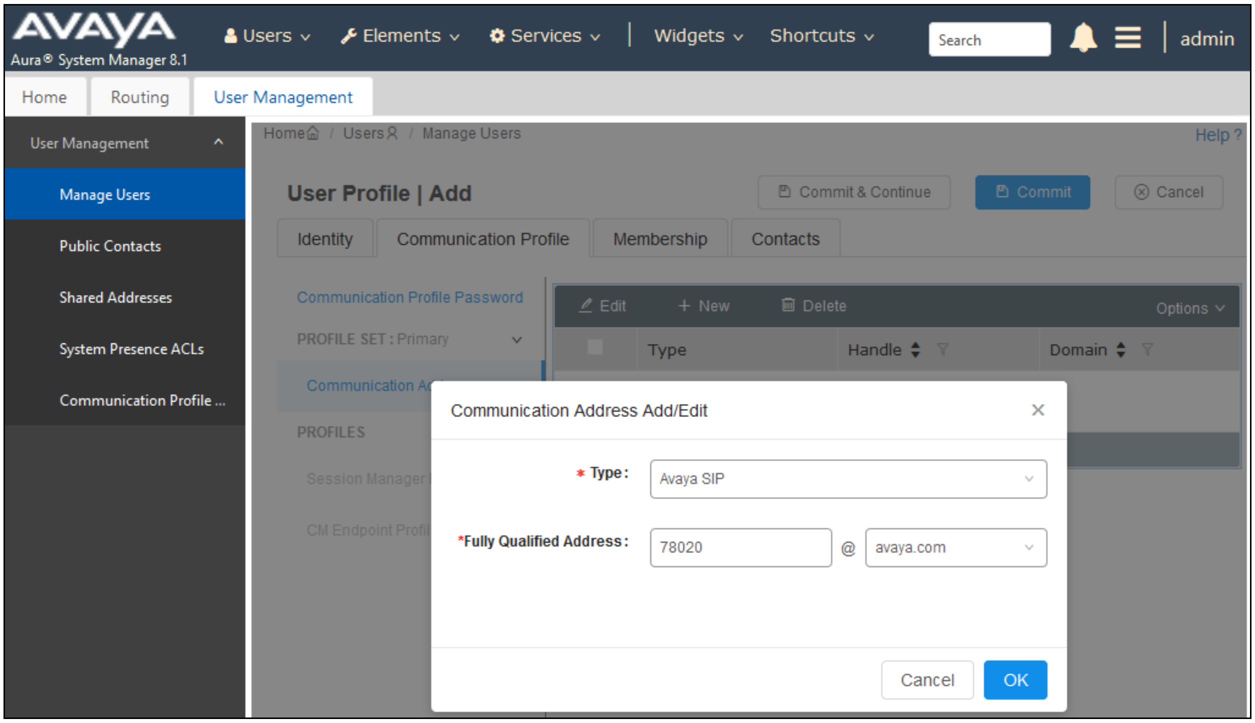
6.3.2. Communication Profile

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



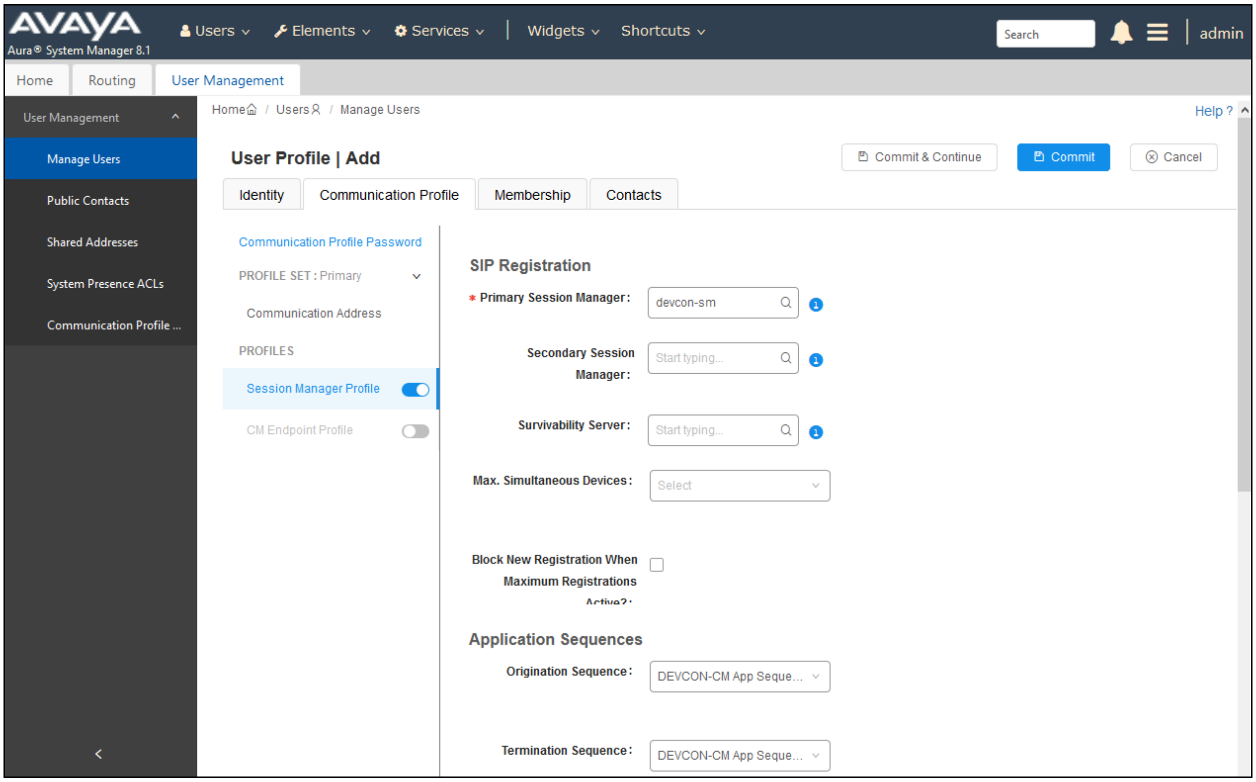
6.3.3. Communication Address

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

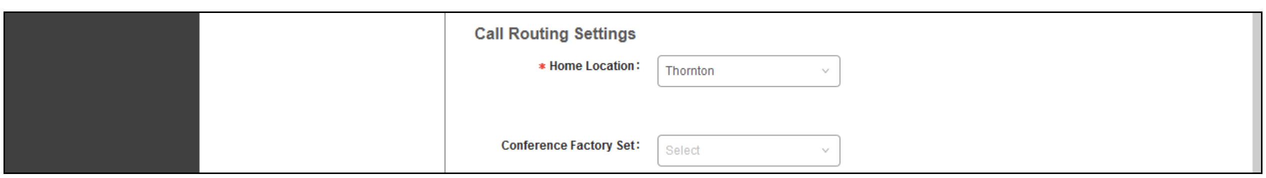


6.3.4. Session Manager Profile

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

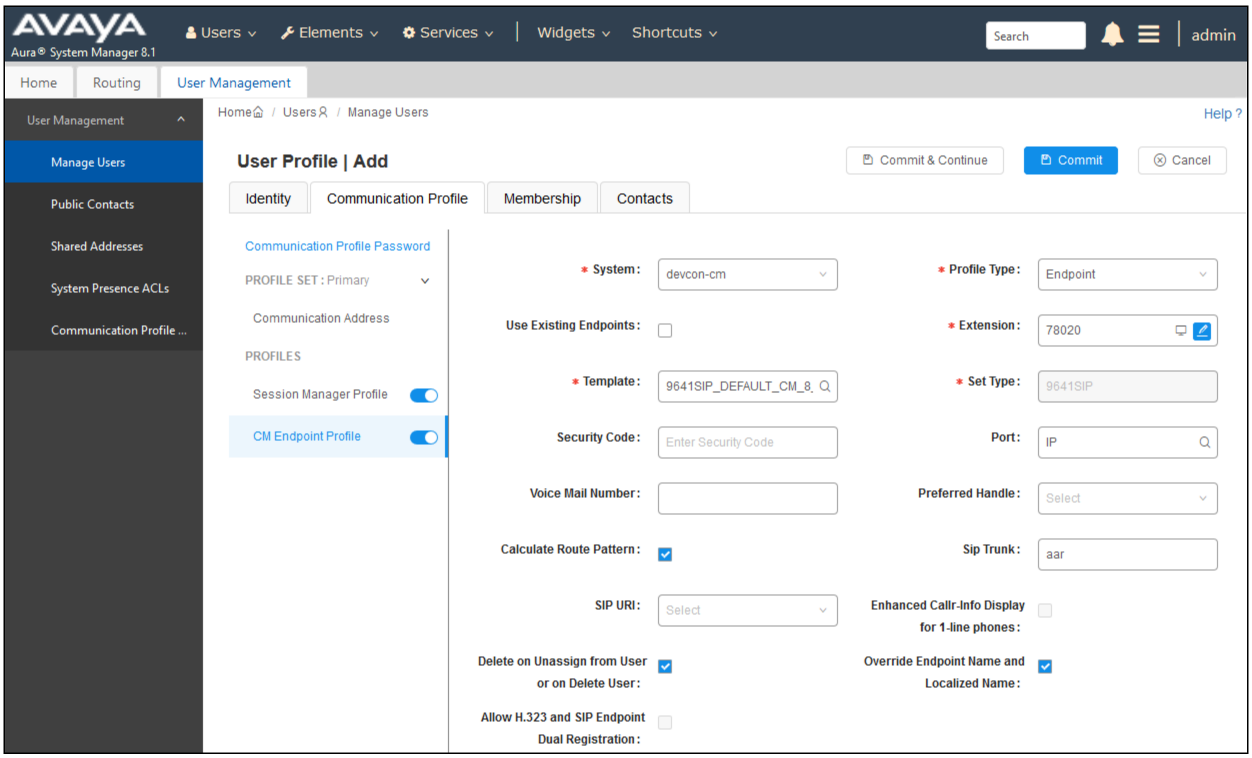


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

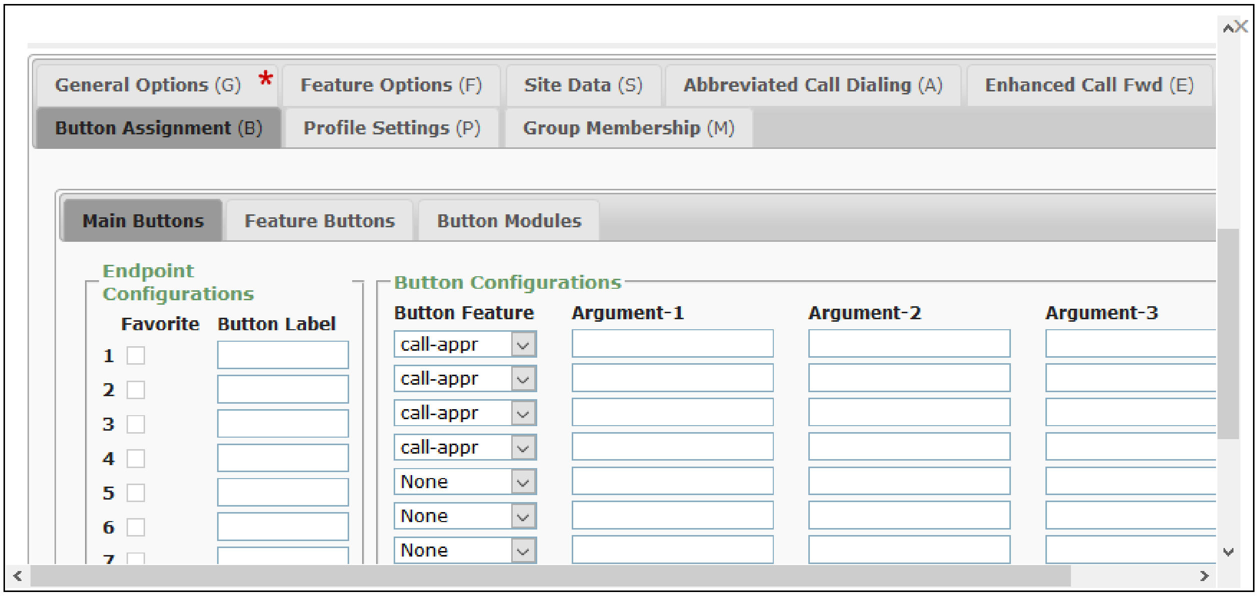


6.3.5. Communication Manager Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.3**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e, Edit icon in **Extension** field) to configure four call appearances in the **Button Assignment** tab.



Navigate to the **Button Assignment** tab and configure four call appearances as shown below. DMP 128 Plus was configured with four call appearances in **Section 7.4**. Click **Done** to return to the previous web page and then **Commit** to save the configuration (not shown).



7. Configure Extron DMP 128 Plus C V

This section provides the procedures for configuring DMP 128 Plus. The procedures fall into the following areas:

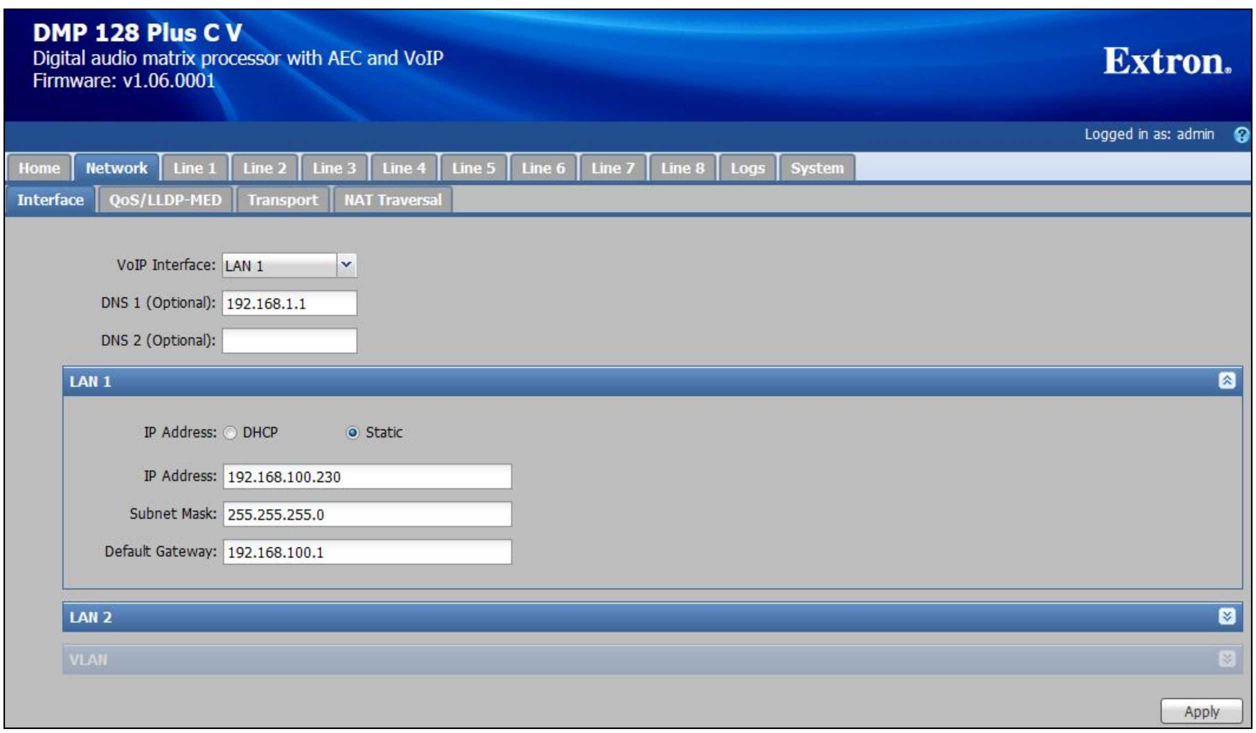
- Launch Web Interface
- Administer Network Settings
- Administer SIP Settings
- Configure the DSP

7.1. Launch Web Interface

DMP 128 Plus was configured via the web interface by using the URL "https://ip-address/www/voip.html" in an Internet browser window, where "ip-address" is the DMP 128 Plus IP address. The web page displayed in the following section is displayed.

7.2. Administer Network Settings

To configure IP network settings, navigate to **Network → Interface** and configure the **LAN 1** settings. For the compliance test, a static IP address, *192.168.100.230*, was assigned to DMP 128 Plus as shown below. Alternatively, DHCP may be used. Click **Apply**.

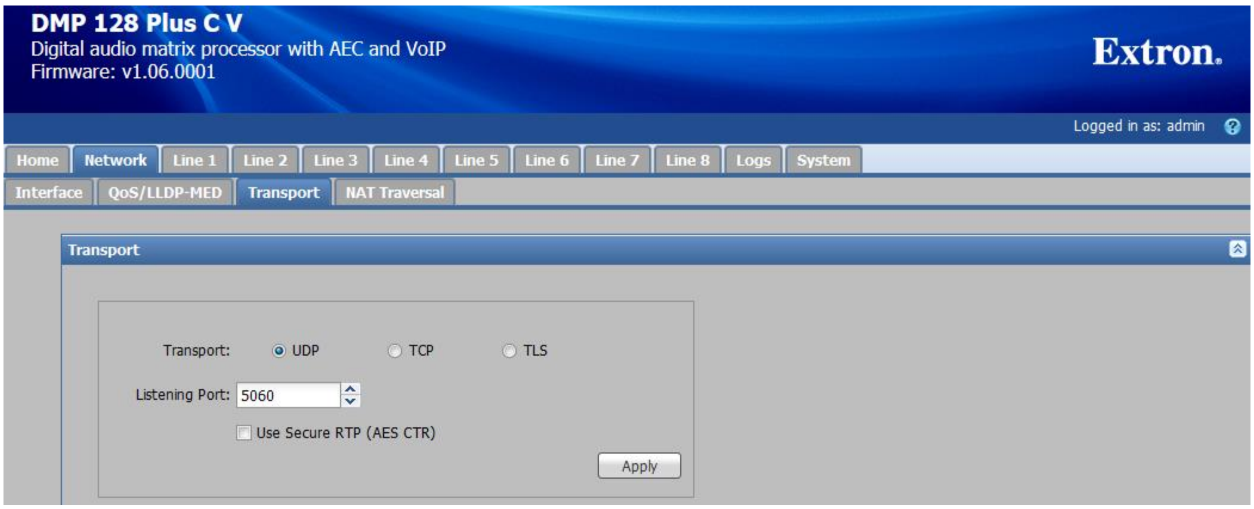


7.3. Administer SIP Settings

To configure SIP settings, select the **Network** tab followed by the **Transport** sub-tab. Click **Configuration** and then select the **SIP** tab. Configure the following fields:

- **Transport** Specify the *UDP* transport protocol.
- **Listening Port** Specify port 5060.
- **Use Secure RTP (AES CTR)** Disable SRTP.

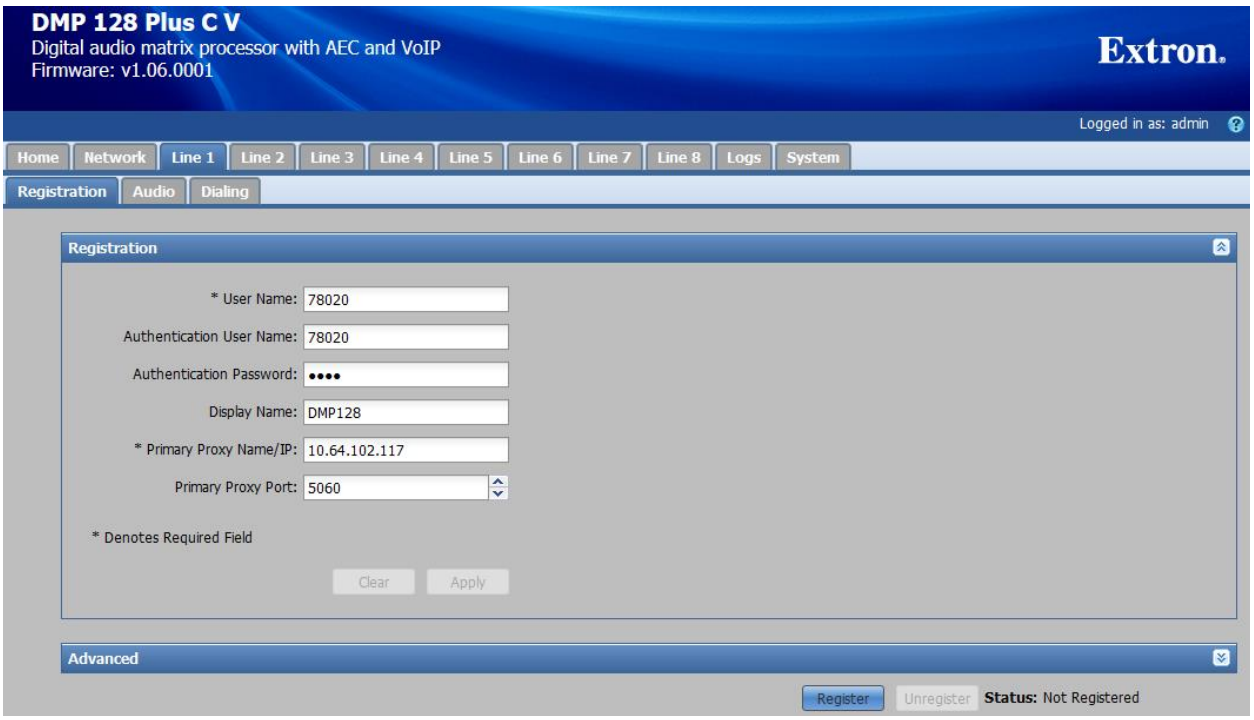
Click **Apply**.



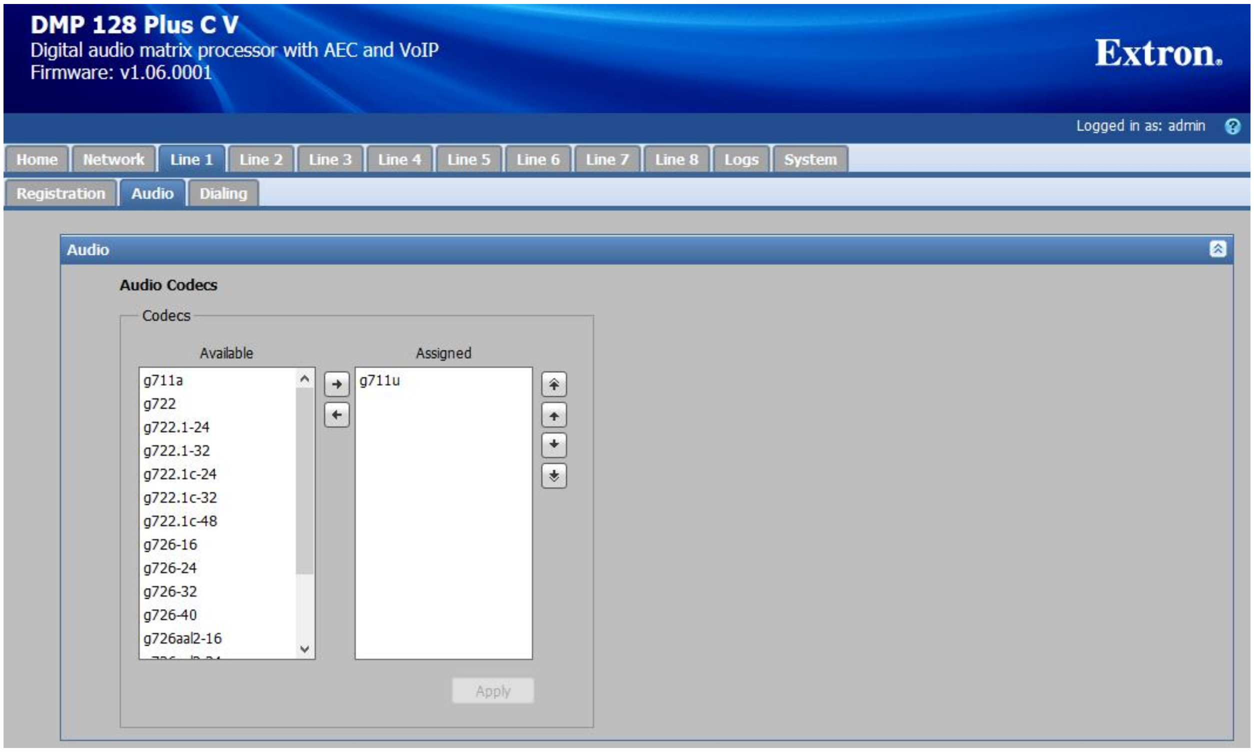
Navigate to **Line 1** tab to configure the SIP registration settings. Configure the following fields:

- **User Name** Specify the SIP extension configured on Session Manager.
- **Authentication User Name** Specify a user name.
- **Authentication Password** Specify the SIP password used to register with Session Manager.
- **Display Name** Provide a display name.
- **Primary Proxy Name/IP** Specify the Session Manager IP address (*10.64.102.117*).
- **Primary Proxy Port** Specify SIP port *5060*.

Click **Apply** to submit the changes and then click the **Register** button to register DMP 128 Plus with Session Manager after all the configuration is in place.

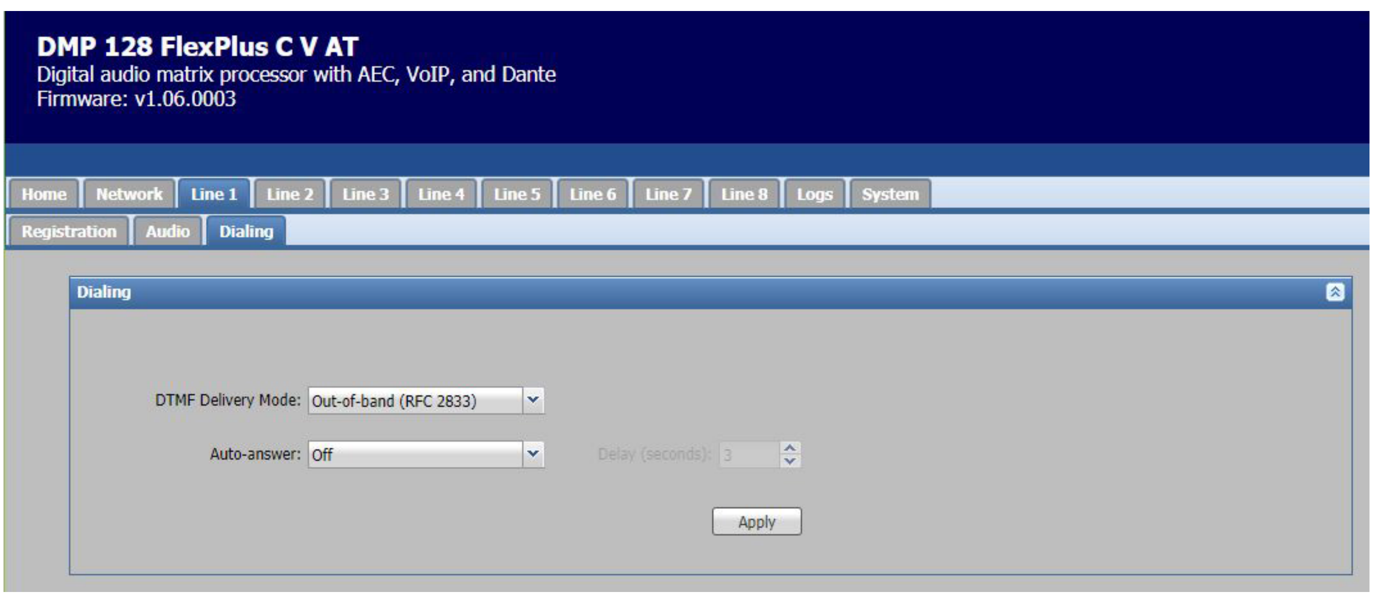


In the **Audio** sub-tab, specify the desired codec, G.711, G.729, or G.722, but not more than one. Refer to the note on codec negotiation in **Section 2.2**. Click **Apply**.



In the **Dialing** sub-tab, accept the default settings shown below.
Set the DTMF Delivery Mode to: **Out-of-Band (RFC 2833)**

Click Apply after select ng **Out of Band – RFC 2833** DTMF signaling method for the line. This can be set per line.



7.3.1 Administer SIP Settings - RTP Port Range

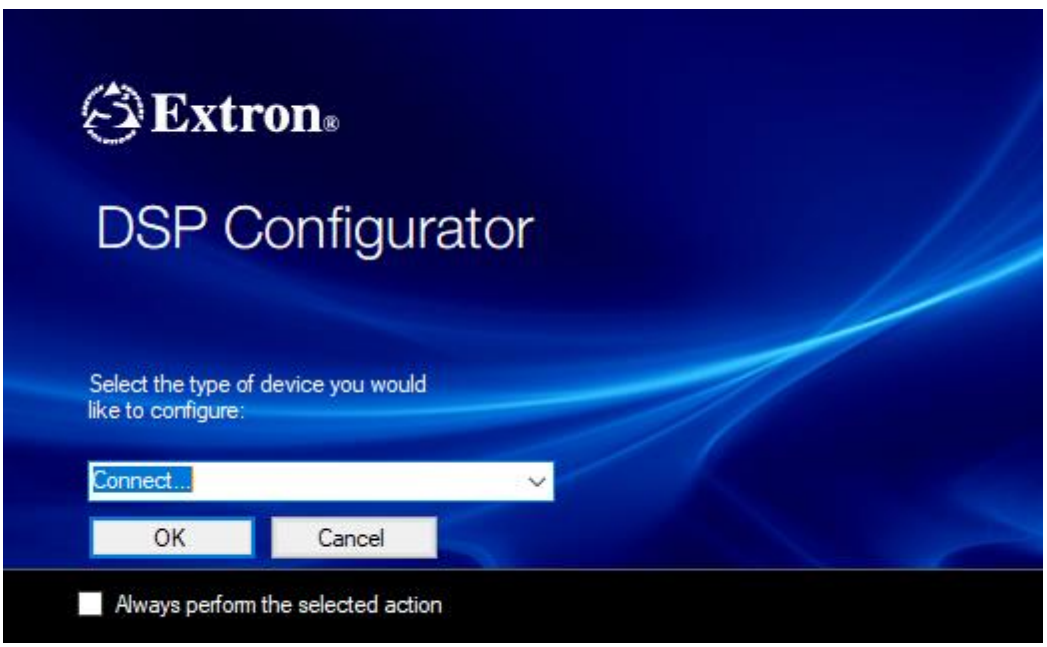
The Default RTP port range of the DMP plus is 50000 - 50099.
Avaya default RTP port range is 1024 - 3329, this may need to be adjusted.
Note: Improper RTP range may result in incoming calls not work properly.

Refer to **Appendix A** in order to adjust this range.

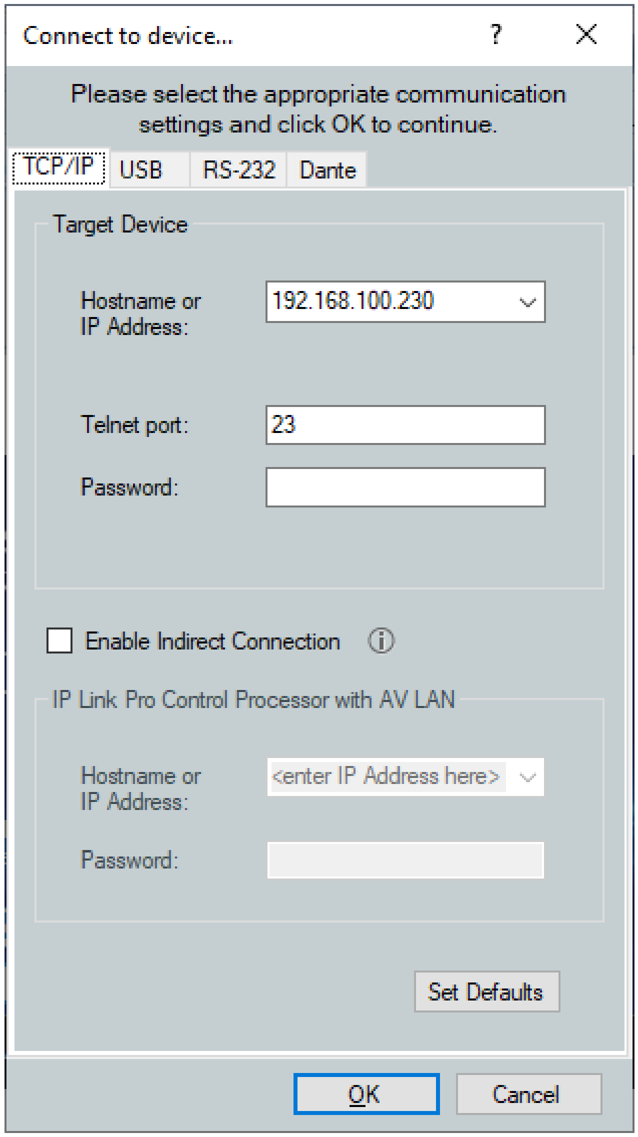
7.4. Configure the DSP

Although the DSP configuration is outside the scope of these Application Notes, the following information is provided for informational purposes only.

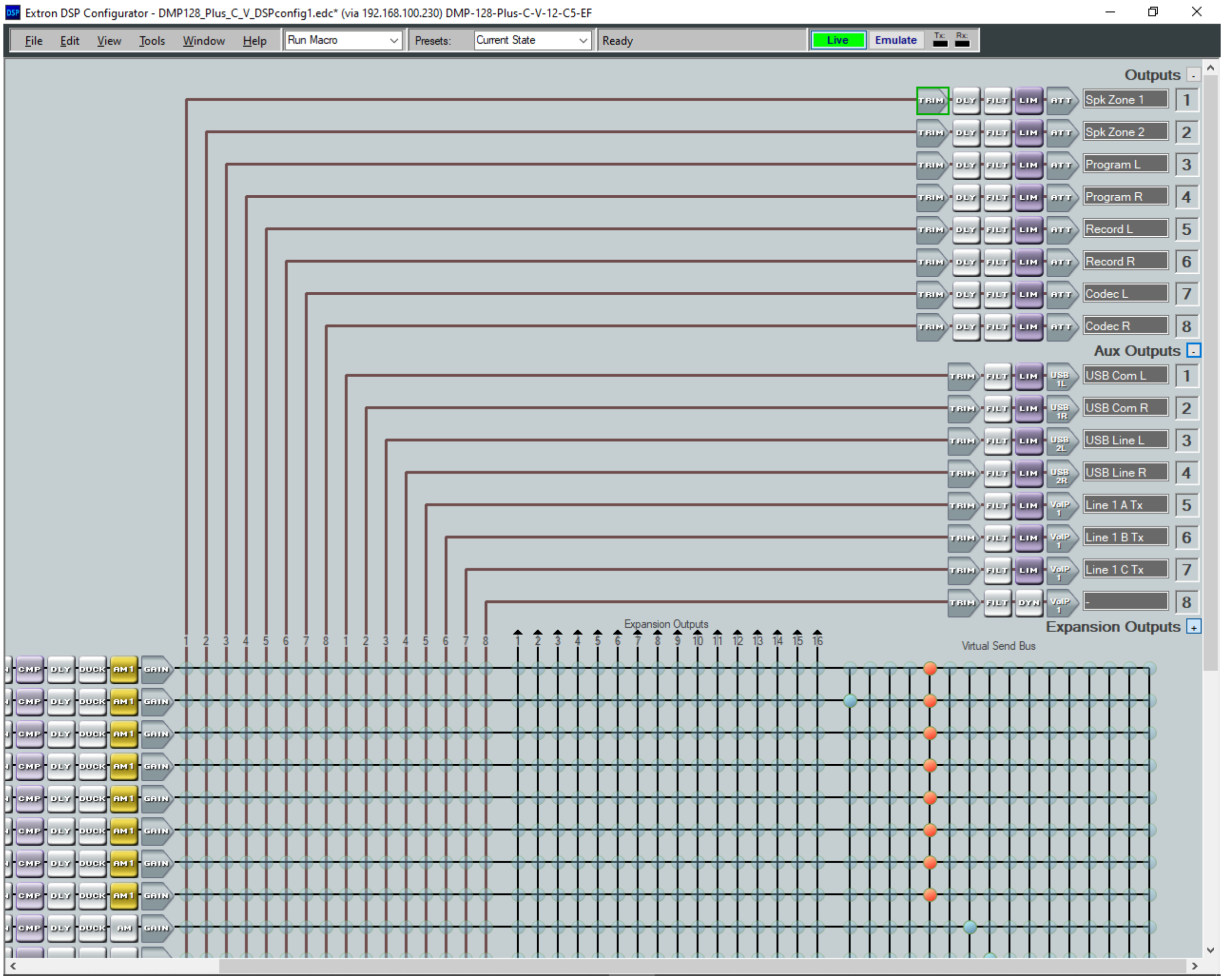
Launch the **DSP Configurator**, select **Connect…** in the drop-down field, and click **OK**.



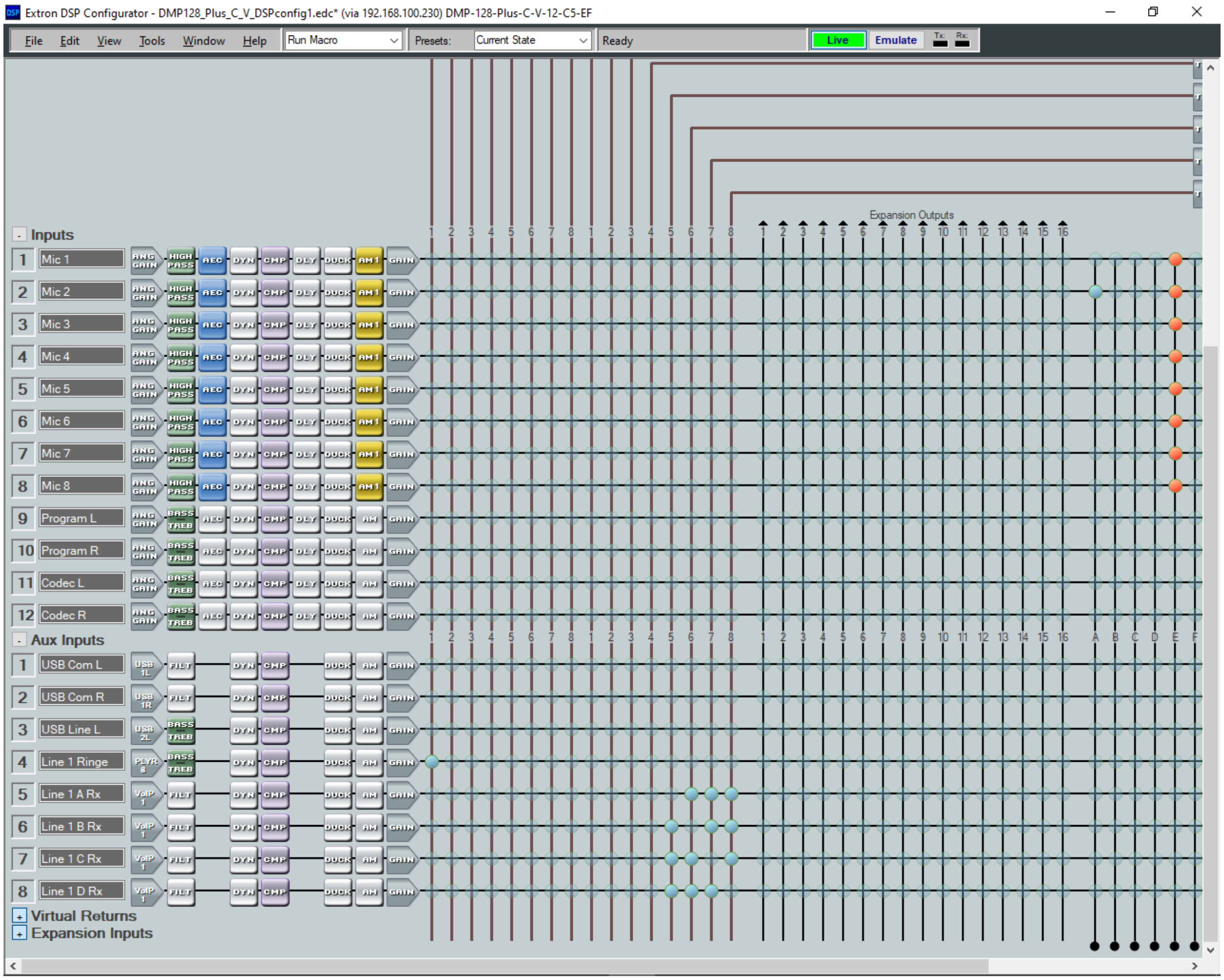
In the **Connect to device…** window shown below, enter the DMP 128 Plus IP address in the **Hostname or IP Address** field and click **OK**.



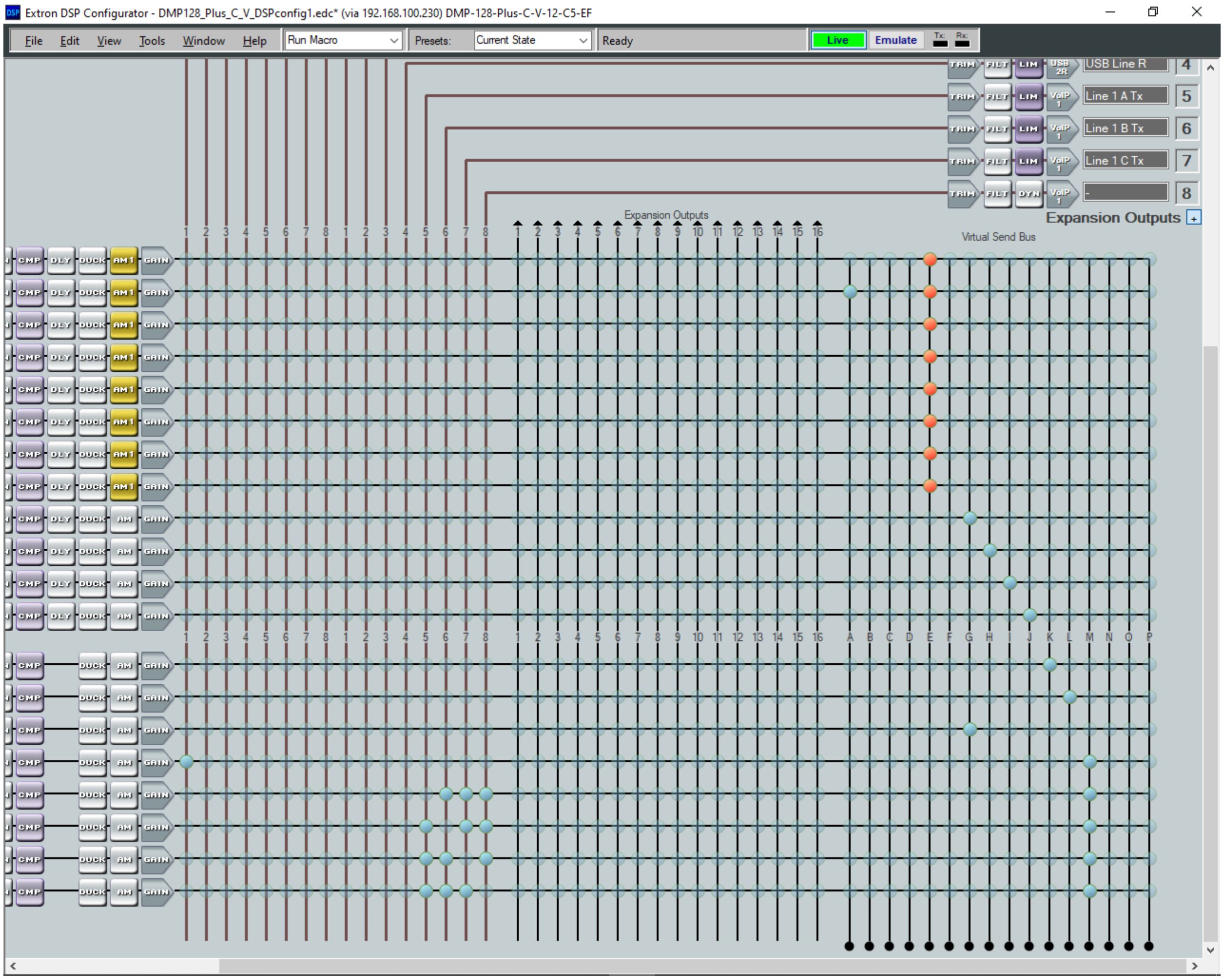
Once connected to DMP 128 Plus, the **DSP Configurator** is displayed as shown below. The following displays the top portion of the screen.



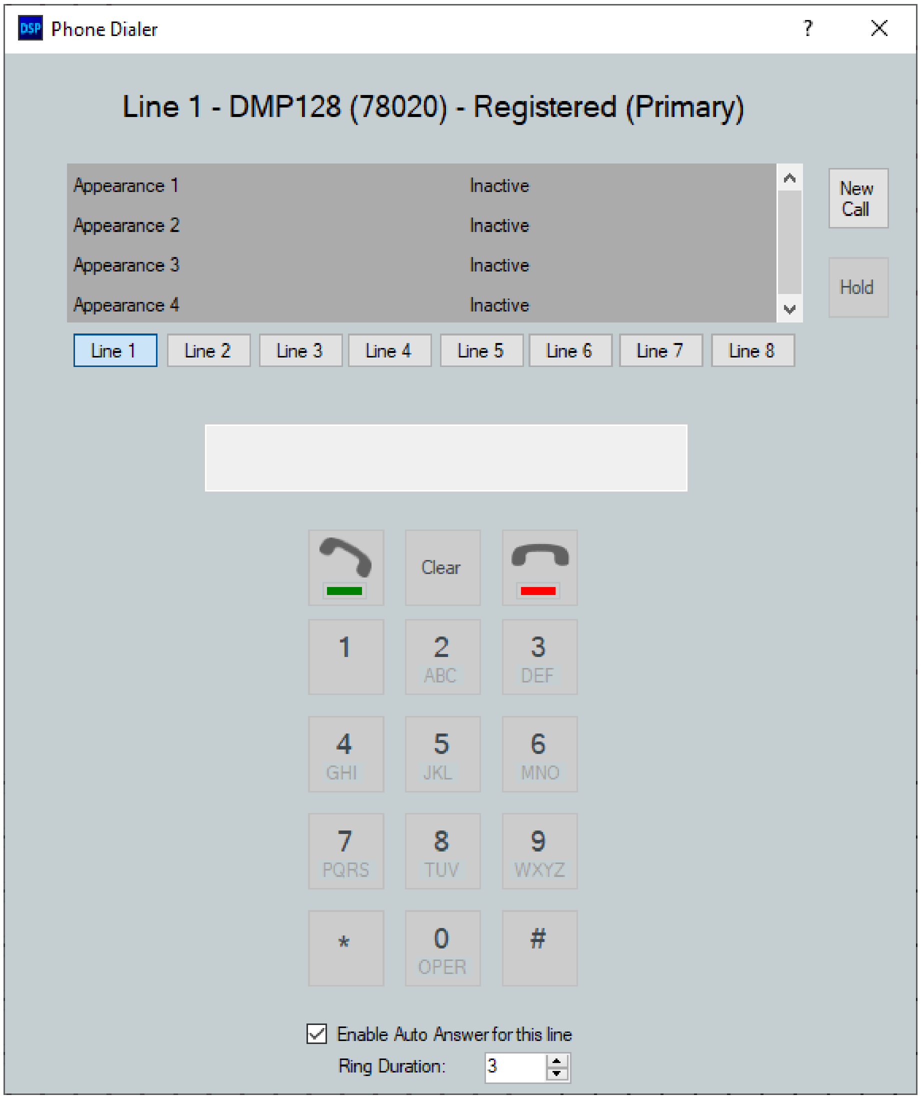
Scrolling down shows that four call appearances were configured for the SIP line and that the audio call appearances were mixed together. This results in all calls being conferenced together automatically.



Scrolling to the right displays the rest of the screen.



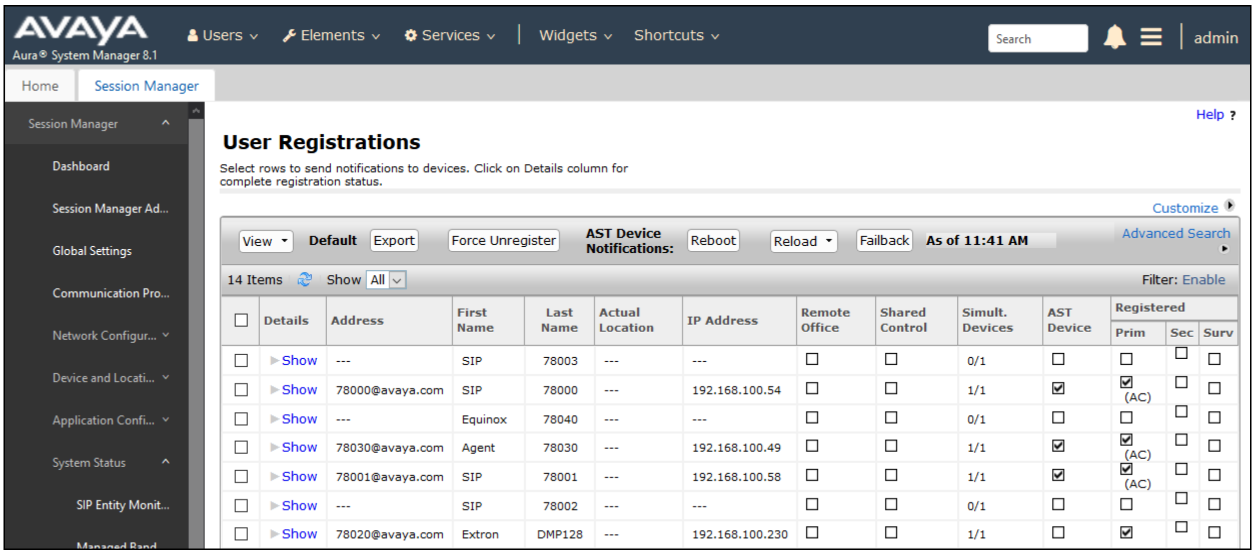
Calls were originated and answered using the **Phone Dialer** accessible from the **DSP Configurator** menu (i.e., **Tools → Phone Dialer**). Typically, the **Extron CCI Pro 700 TouchLink Pro Conference Room Control Interface** or another Extron customizable touchpanel device would be used by customers for this purpose, which would provide audio tone feedback for each call.



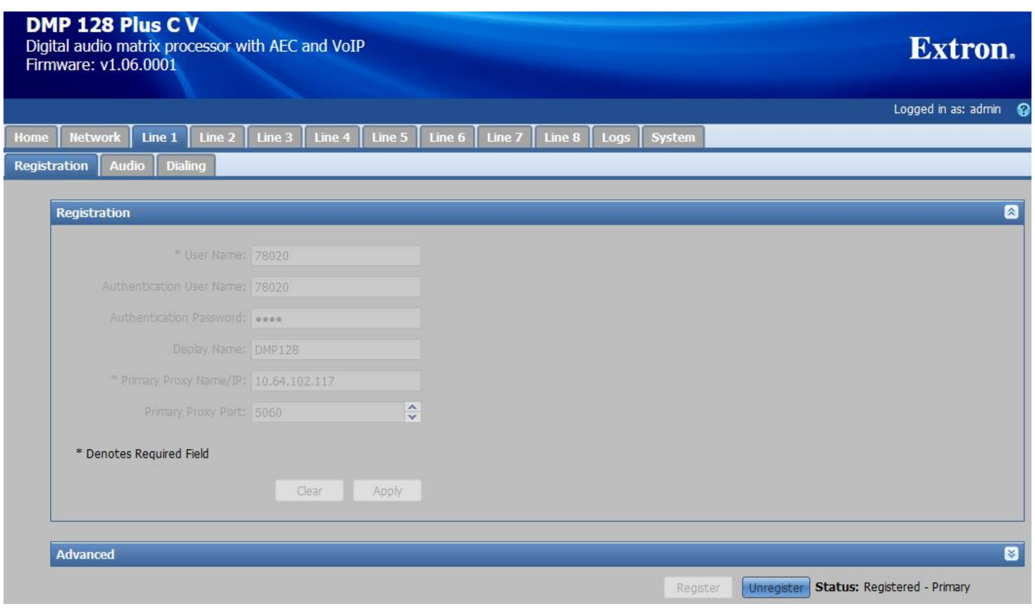
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and DMP 128 Plus C V.

1. Verify that DMP 128 Plus has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.



2. Alternatively, the registration status may be viewed on the DMP 128 Plus web interface in the **Registration** tab or in the **Phone Dialer** shown on the next page.



3. Verify basic telephony features by establishing calls between DMP 128 Plus and local phones.

9. Conclusion

These Application Notes described the configuration steps required to integrate Extron DMP 128 Plus C V with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Extron DMP 128 Plus C V was able to establish calls with H.323 stations, SIP stations, and the PSTN. In addition, basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. Additional References

This section references the Avaya and Extron documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com and the Extron documentation is available at https://www.extron.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 6, March 2020.
[2] *Administering Avaya Aura® System Manager 8.1.x*, Issue 5, March 2020.
[3] *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 5, March 2020.
[4] *Extron DMP 128 Plus User Guide*, 68-2826-01 Rev. H, 12 19.
[5] *Extron DMP 128 Plus Series Avaya Configuration Guide*, Revision 1.2, March 27, 2020.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.

11.0 Troubleshooting

In the event of failure to register, review the following:

- Check that the credentials specified as part of the AVAYA setup are correctly entered into the registration fields for each line.

- Check network interface settings, including DNS fields (particularly if a proxy domain name is being used rather than an IP address).

- Verify the RTP port range has been set correctly
  - Avaya default is 1024 – 3329, Refer to Appendix A in order to adjust this range.

- Check to audio codecs, make sure **<u>only one</u>** Codec is assigned (g711u or g 711a)

- Click on the **Logs** tab to inbound and outbound SIP transactions. The absence of inbound transactions indicates a network routing problem. Registration-specific problems may be indicated by corresponding SIP responses such as *403 – Forbidden*.

- If the Logs indicate a registration failure, edit the Avaya account so that User ID and Digest User are the same as the Directory Number. Then, update both the User Name **and** Authentication Name fields on the DMP Plus Series with the Directory Number. Although this is not a prerequisite for registration, it may assist in identifying configuration issues.

12.0 Configuration File – Attached to PDF

If needed, the configuration file “voipConfig.conf” is attached to the PDF

- To access the file select “Attachments”  from the left side bar – see figure A1
- Then save the attachment, before uploading to DMP Plus – see Figure A2 below

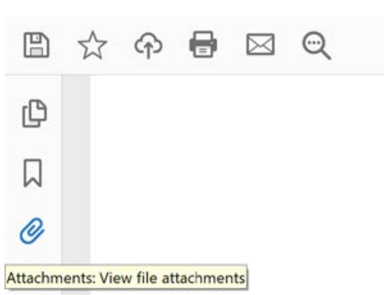


Figure A1 Save Attachment

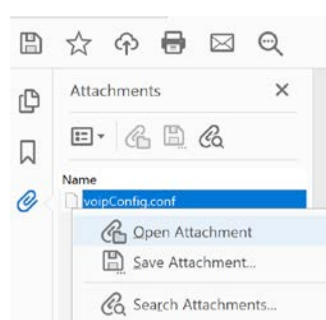


Figure A2 Show Attachments

Appendix A: RTP Port Range

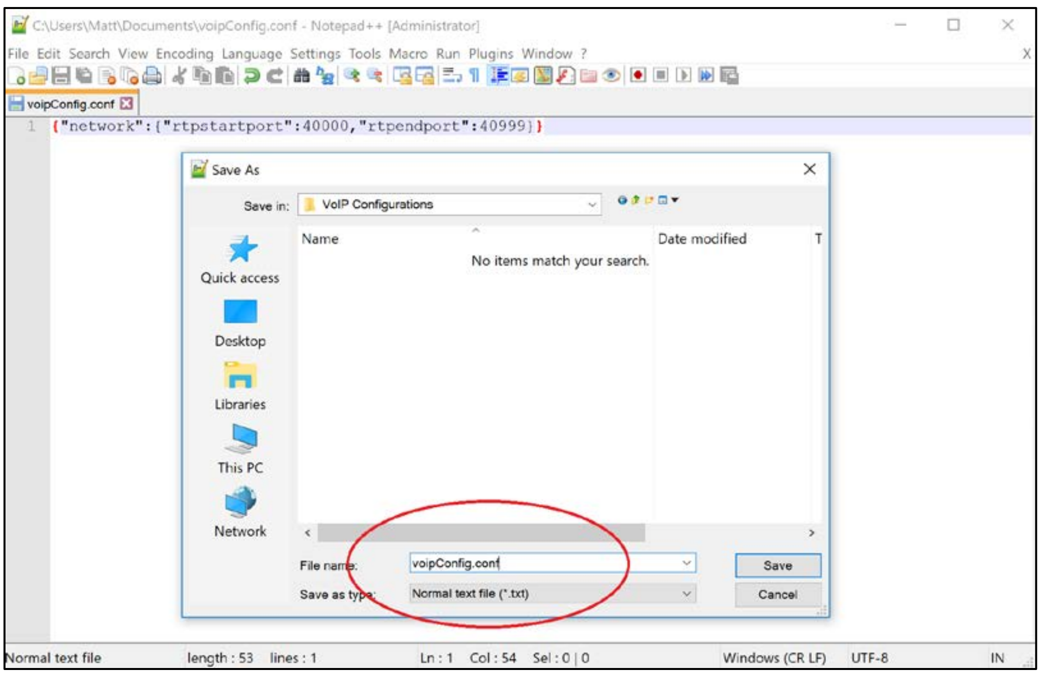
The default port range for VoIP RTP traffic on the DMP Plus Series is **50000 – 50999**. To change this range, the following steps must be carried out.
Avaya default RTP port range is 2048 - 3329

1. Create a new blank text file using a suitable basic text editor.
    a. Example "**voipConfig.conf**"is attached to this PDF, see **Section 12.0**

2. Enter the following text into the document (in this example, the port range is being changed to 2048 - 3329; replace these values with the desired range) -

```
{"network":{"rtpstartport":2048,"rtpendport":3329}}
```

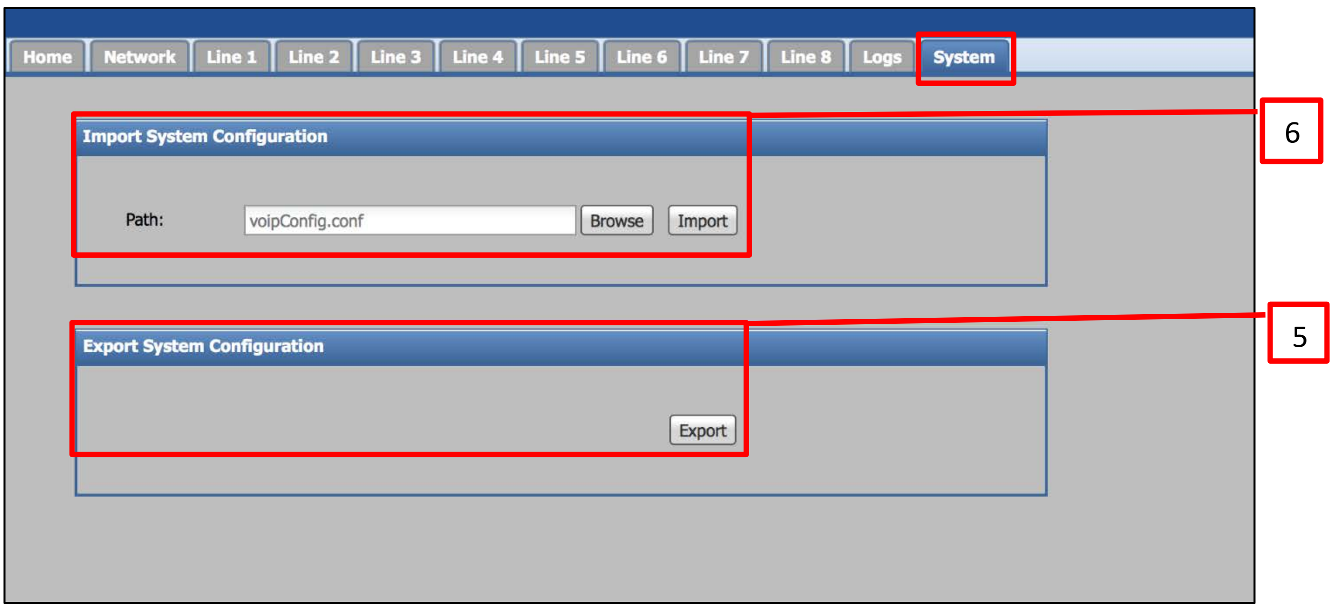
3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP Plus Series VoIP configuration webpage and click on the **System** tab.

5. Under **Export System Configuration**, click the **Export** button to back up the current VoIP configuration to disk. The file will be saved in the default web browser download directory.

6. Under **Import System Configuration**, click the **Browse** button to locate the **voipConfig.conf** file created in steps 1 to 3.



7. Click the **Import** button to update the DMP Plus Series with the new RTP Port Range settings. A notification will appear once the settings have applied successfully.

Appendix B: Outgoing Call Termination Mode

Some call server configurations require the use of a CANCEL SIP message to terminate unanswered outgoing calls, rather than a traditional BYE message. This requirement will become apparent if far end devices continue to ring after the DMP Plus Series has ended the call.

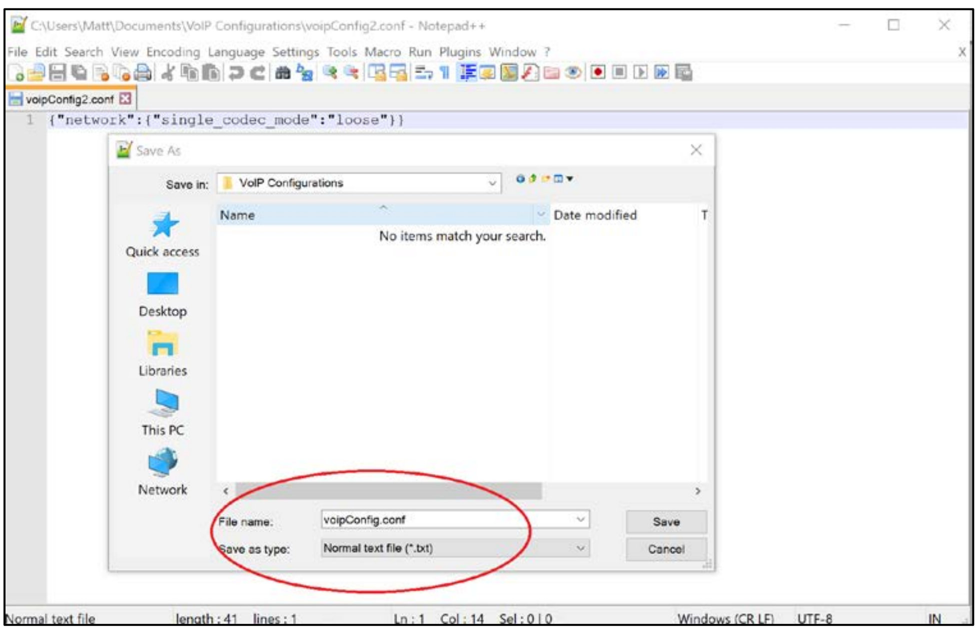
To change the outgoing call terminate mode from BYE to CANCEL (default is BYE), the following steps must be carried out. **Requires FW 1.01.0007-b004 or later.**

1. Create a new blank text file using a suitable basic text editor
   a. Example “**voipConfig.conf**”is attached to this PDF, see **Section 12.0**

2. Enter the following text into the document -

```
{"network":{"sip_outgoing_termination_use_cancel_enable":true}}
```

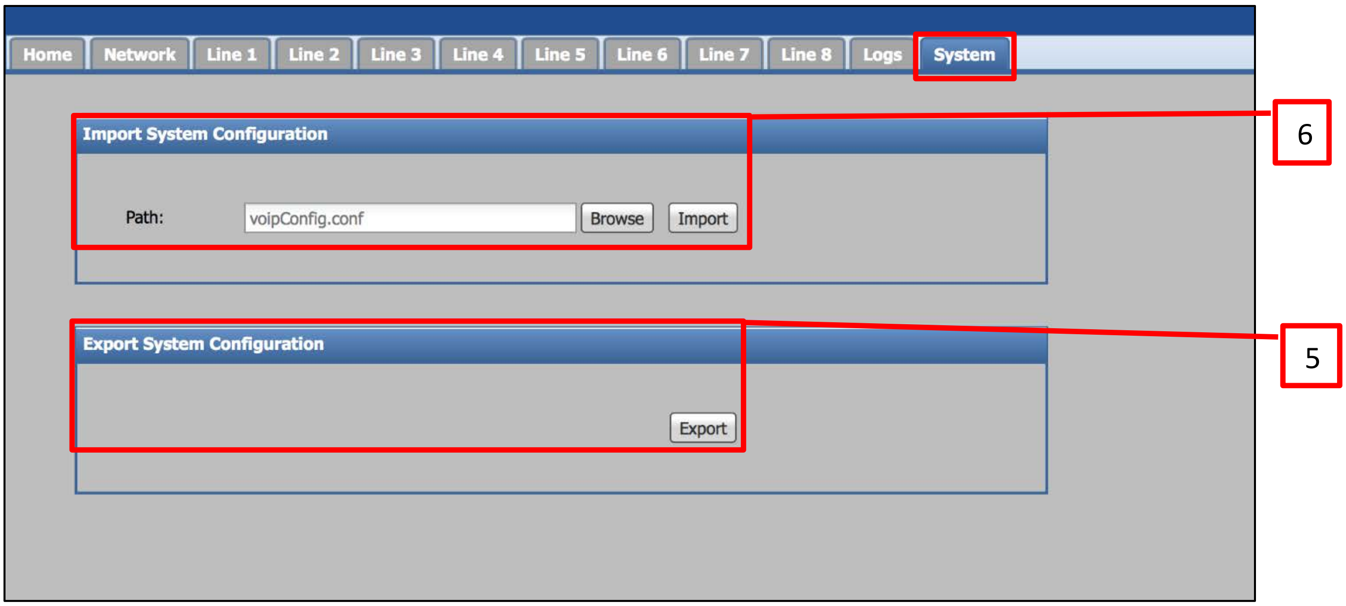
3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP Plus Series VoIP configuration webpage and click on the **System** tab.

5. Under **Export System Configuration**, click the **Export** button in order to back up the current VoIP configuration to disk. The file will be saved in the default web browser download directory.

6. Under **Import System Configuration**, click the **Browse** button to locate the **voipConfig.conf** file created in steps 1 to 3.



Click the **Import** button to update the DMP Plus Series with the new settings. A notification will appear once the settings have applied successfully.

To return to BYE mode, send the following string using the same method:

```
{"network":{"sip_outgoing_termination_use_cancel_enable":false}}
```

Appendix C: Automatic Line Re-Registration

Some call managers and networks go into maintenance windows which do not allow VoIP endpoints to register or maintain their registration. To help resolve this issue the Automatic Line Re-Registration function can be configured to re-register a line if line registration is unexpectedly lost. This function causes the VoIP interface to re-attempt a line re-registration if the first automatic re-registration attempt fails.

In order to use this feature, the line must first be registered to the call manager.
Note: When enabled, this function will attempt re-registration once the SIP timer has expired. By default the SIP timer is set to 3600 seconds (60 mins). By default, the Automatic Line Re-Registration feature is disabled, with the "registration_fail_retry_count" set to zero (0).

To set up Automatic Line Re-Registration, the following steps must be carried out. **Requires FW 1.02.0001-b001 or later.**

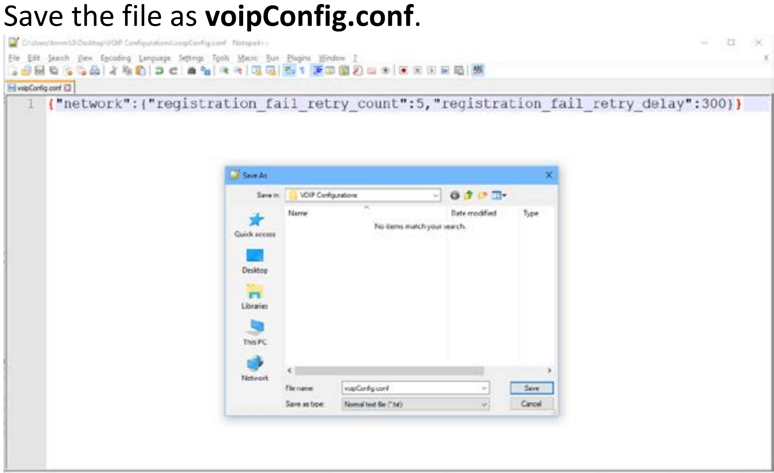
1. Create a new blank text file using a suitable basic text editor
   a. Example "**voipConfig.conf**"is attached to this PDF, see **Section 12.0**

2. Enter the following text into the document –

```
{"network":{"registration_fail_retry_count":5,"registration_fail_retry_delay":300}}
```

   a. registration_fail_retry_count":5
      This is the number of attempts a Line will make to re-register
      i. Example above is set to five (5) reconnections attempts
      ii. If this is set to zero (0), the feature is disabled
      iii. Valid Range of values: 0 - 99
   b. registration_fail_retry_delay":300
      Amount time between registration attempts in seconds
      i. Example above is set to 300 seconds (5 mins) between reconnections attempts
      ii. Valid Range of values: 120 - 3600

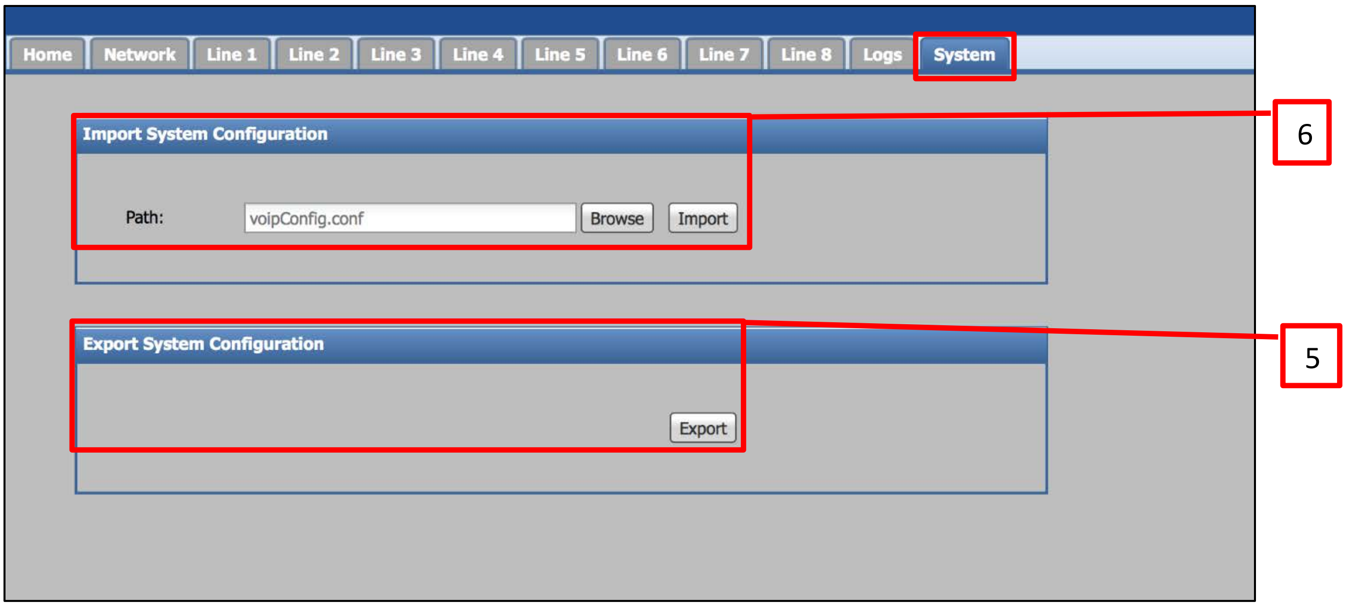
3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP Plus Series VoIP configuration webpage and click on the **System** tab.

5. Under **Export System Configuration**, click the **Export** button in order to back up the current VoIP configuration to disk. The file will be saved in the default web browser download directory.

6. Under **Import System Configuration**, click the **Browse** button to locate the **voipConfig.conf** file created in steps 1 to 3.



Click the **Import** button to update the DMP Plus Series with the new settings. A notification will appear once the settings have applied successfully.

To disable to Auto-Reregistration mode, send the following string using the same method:

{"network":{"registration_fail_retry_count":0,"registration_fail_retry_delay":200}}

Extron

Avaya Devconnect

December 10, 2019

Declaration of conformance for Extron DMP Plus Series

We, Extron herby confirms that the following DMP plus products:

- DMP 128 Plus C V
- DMP 128 Plus C V AT
- DMP 128 FlexPlus C V AT
- DMP 64 Plus C V
- DMP 64 Plus C V AT

Are based on the same platform and therefore:

- Use identical SIP stack
- Use the same firmware version

The differences in the DMP plus models:

- DMP 128 Plus C V (AT)
    - Supports 12 mic/line inputs and 8 line outputs
    - Supports 12 channels of Acoustical Echo Cancelation (AEC)

- DMP 128 FlexPlus C V AT
    - Supports 4 mic/line inputs and 8 line outputs
    - Supports 12 channels of Acoustical Echo Cancelation (AEC)

- DMP 64 Plus C V (AT)
    - Supports 6 mic/line inputs and 4 line outputs
    - Supports 6 channels of Acoustical Echo Cancelation (AEC)

- Models ending in "AT" support Dante-equipped products that provide scalable audio transport over a local area network using standard Internet protocols.

Best regards

David Pincek
VP Product Development

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