Avaya Solution & Interoperability Test Lab

Application Notes for Extron DMP 128 Plus C V with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1 - 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. The Extron DSP Configurator 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. VoIP configuration is performed via a dedicated webpage, simplifying the setup and management for IT personnel. 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.

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. The Extron DSP Configurator 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. VoIP configuration is performed via a dedicated webpage, simplifying the setup and management for IT personnel. Extron DMP 128 Plus C V registers to Avaya Aura® Session Manager as a SIP endpoint.

For this compliance test, DMP 128 Plus uses the TLS transport protocol to secure SIP messages. During the TLS handshake with Session Manager, DMP 128 Plus uses the Server Name Indication (SNI) extension of TLS in the Client Hello message. The SNI must be the Session Manager FQDN specified in the TLS certificate, not the IP address of the Session Manager SIP interface. As a result, DMP 128 Plus uses the SNI as the SIP domain for registration and outgoing calls. However, it will also accept other SIP domains for incoming calls when Session Manager is properly configured. For example, the SIP user for DMP 128 Plus was configured with two SIP domains in the Communication Addresses, one using the SNI (e.g., *devcon-sm.avaya.com*) as the SIP domain and another one using an alternate SIP domain (e.g., *avaya.com*).

For the compliance test, Communication Manager only used an existing SIP domain (e.g., *avaya.com*), which was already in use by existing SIP endpoints. It was not configured to use the Session Manager FQDN (e.g., *devcon-sm.avaya.com*) as a SIP domain, which was only used by DMP 128 Plus in this configuration. Therefore, Communication Manager sent the existing SIP domain (e.g., *avaya.com*) for all calls, including calls to DMP 128 Plus. However, Session Manager delivered the call to DMP 128 Plus, because the SIP user was configured with two different SIP domains as mentioned earlier.

Alternatively, Communication Manager could have been configured with another set of IP network region, IP codec set, signaling group, and trunk group associated with the SIP domain used by DMP 128 Plus. This approach would require SIP trunk ports to be allocated specifically for DMP 128 Plus calls and media resources to be assigned to the new IP network region.

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 deskphones and the PSTN, and exercising basic telephony features, such as hold, mute, blind transfer and conference. Additional telephony features, such as call forward, coverage, 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 utilized TLS/SRTP.

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.
- TLS transport protocol.
- SRTP media encryption.
- Support of G.711 and G.729 codecs.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold/resume, mute/unmute, redial, multiple calls, call forwarding, blind transfer, conference using audio mixing, and long duration calls.

- Extended telephony features using Communication Manager FACs for Call Forward, Coverage, Call Park/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 TouchLink Pro, a customizable touch panel, 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 does not support receiving a re-INVITE without SDP, which could have adverse effects on shuffled calls and various hold scenarios.
- DMP 128 Plus supports blind transfers. However, blind transfer was performed via a command via a Telnet session because transfers cannot be initiated from the Phone Dialer tool.
- 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/contactform.aspx?action=techsupport

3. Reference Configuration

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

- Avaya Aura® Communication Manager with an Avaya Media Gateway.
- Media resources in the Avaya 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 endpoints.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J100 Series SIP Phones.
- Extron DMP 128 Plus C V and 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.

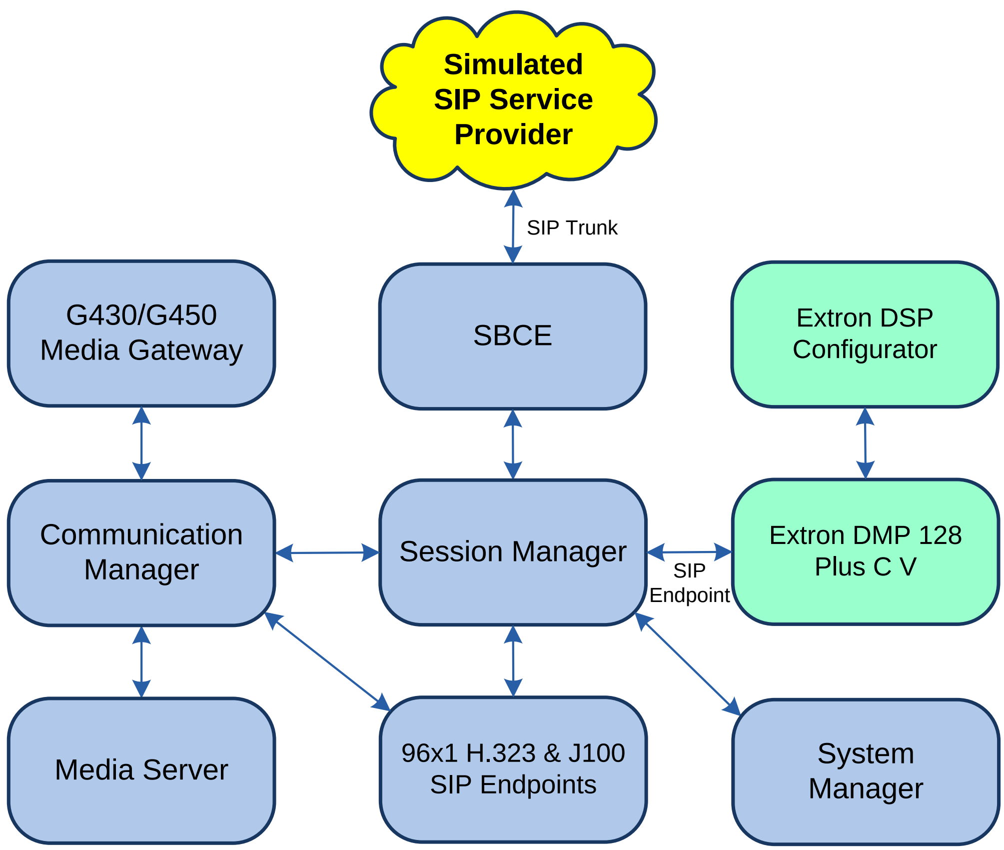


Figure 1: Avaya SIP-based Network with Extron DMP 128 Plus C V

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	10.1.2.0.0-FP2
Avaya G430 Media Gateway	FW 42.8.0
Avaya G450 Media Gateway	FW 42.7.0
Avaya Aura® Media Server	v.10.1.0.77
Avaya Aura® System Manager	10.1.2.0 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.2.0.0-071476 Feature Pack 2
Avaya Aura® Session Manager	10.1.2.0.1012016
Avaya Session Border Controller for Enterprise	10.1.1.0-35-21872
Avaya 96x1 Series IP Deskphones	6.8.5.3.2 (H.323)
Avaya J100 Series SIP Phones	4.0.13.0.6
Extron DMP 128 Plus C V	1.08.0005-b002
Extron DSP Configurator	2.27.0.42

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 Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

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. The SIP station configuration for Extron DMP 128 Plus C V is configured through Avaya Aura® System Manager in **Section 6.2**.

5.1. Verify Communication Manager 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: V20                                Software Package: Enterprise
         Location: 2                                   System ID (SID): 1
         Platform: 28                                  Module ID (MID): 1

                                                                USED
                            Platform Maximum Ports:  48000    131
                                  Maximum Stations:  36000     37
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:  41000      0
                Maximum Off-PBX Telephones -   OPS:  41000     23
                Maximum Off-PBX Telephones - PBFMC:  41000      0
                Maximum Off-PBX Telephones - PVFMC:  41000      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                Maximum Off-PBX Telephones -   EMX:  36000      0
                     Maximum Survivable Processors:  313        0



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

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. 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 used by existing SIP users on Session Manager. In this configuration, the domain name is *avaya.com*. **IP-IP Direct Audio** (shuffling) should be disabled as mentioned in **Section 2.2**. 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: DMP 128 Plus uses the Session Manager FQDN (e.g., *devcon-sm.avaya.com*) as the SIP domain for registration and outgoing calls. However, DMP 128 Plus also accepts incoming calls using the *avaya.com* domain, which is used by other SIP endpoints. This is accomplished by specifying multiple SIP domains in the Communication Addresses of the SIP user configured in **Section 6.5.3**,

For this compliance test, another set of IP network region, IP codec set, signaling group, and trunk group was not configured for *devcon-sm.avaya.com*, although this is an option. Instead, Communication Manager used *avaya.com*, as configured in this IP network region, for all calls, including calls to DMP 128 Plus.

```
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.711 and G.729 codecs. The following IP Codec Set is shown configured with the G.711 codec.

DMP 128 Plus supports *1-srtp-aescm128-hmac80*. However, the IP codec should not include RTP (i.e., *none*) under **Media Encryption. Encrypted SRTCP** may be left at *best-effort*.

```
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: 2-srtp-aescm128-hmac32
 3:
 4:
 5:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*. DMP 128 Plus will received calls with this domain.
- **Direct IP-IP Audio Connections** could be disabled on this form or in the IP network region form as it was for the compliance test.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? n        Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to DMP 128 Plus, Avaya SIP deskphones, and Avaya Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure default values for the remaining fields.

```
add trunk-group 10                                                  Page   1 of  5
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm                COR: 1        TN: 1        TAC: 1010
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern 10 as shown below.

```
change aar analysis 78                                                 Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    78                    5    5      10        lev0         n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                                Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 10   0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

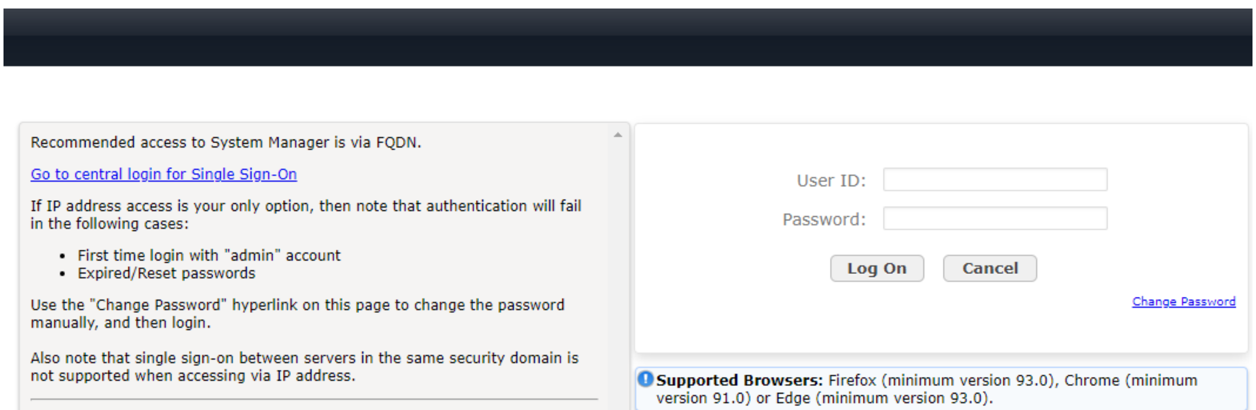
This section covers Session Manager configuration, including the following areas:

- Launch System Manager
- Administer Domains
- Administer Locations
- Administer Session Manager SIP Entity, including SIP FQDN, Location, and Transport Protocol for DMP 128 Plus
- Administer SIP User
- Verify Session Manager FQDN

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.

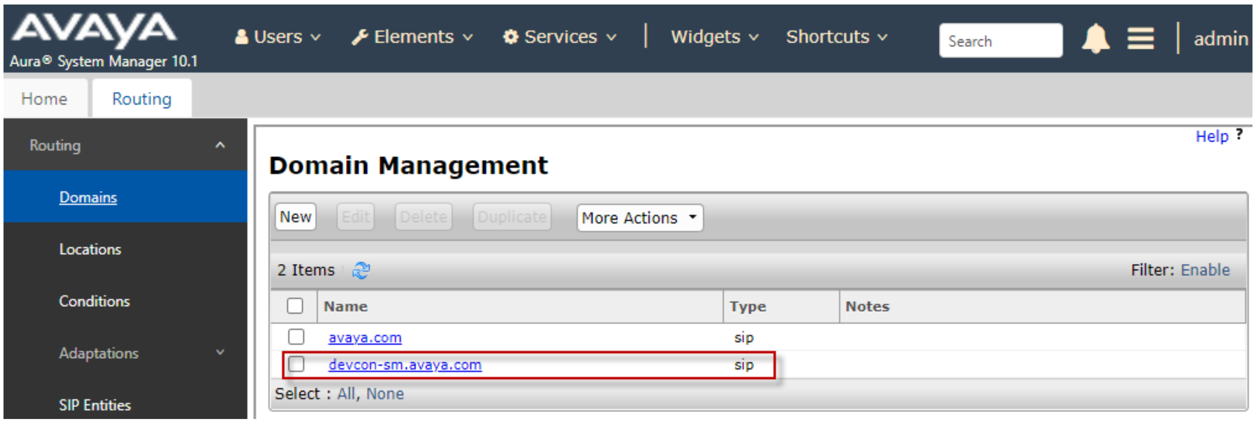
6.1. Launch System Manager

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.



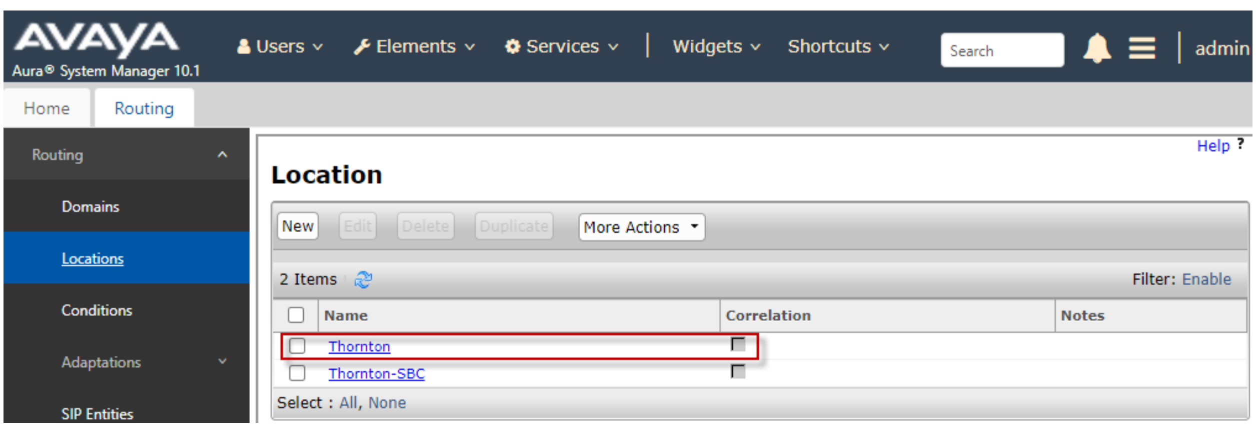
6.2. Administer Domains

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by navigating to **Elements → Routing → Domains.** Add the SIP domain used by DMP 128 Plus. For this compliance test, DMP 128 Plus used *devcon-sm.avaya.com*. The avaya.com domain shown below was used by all other SIP endpoints during the compliance test.



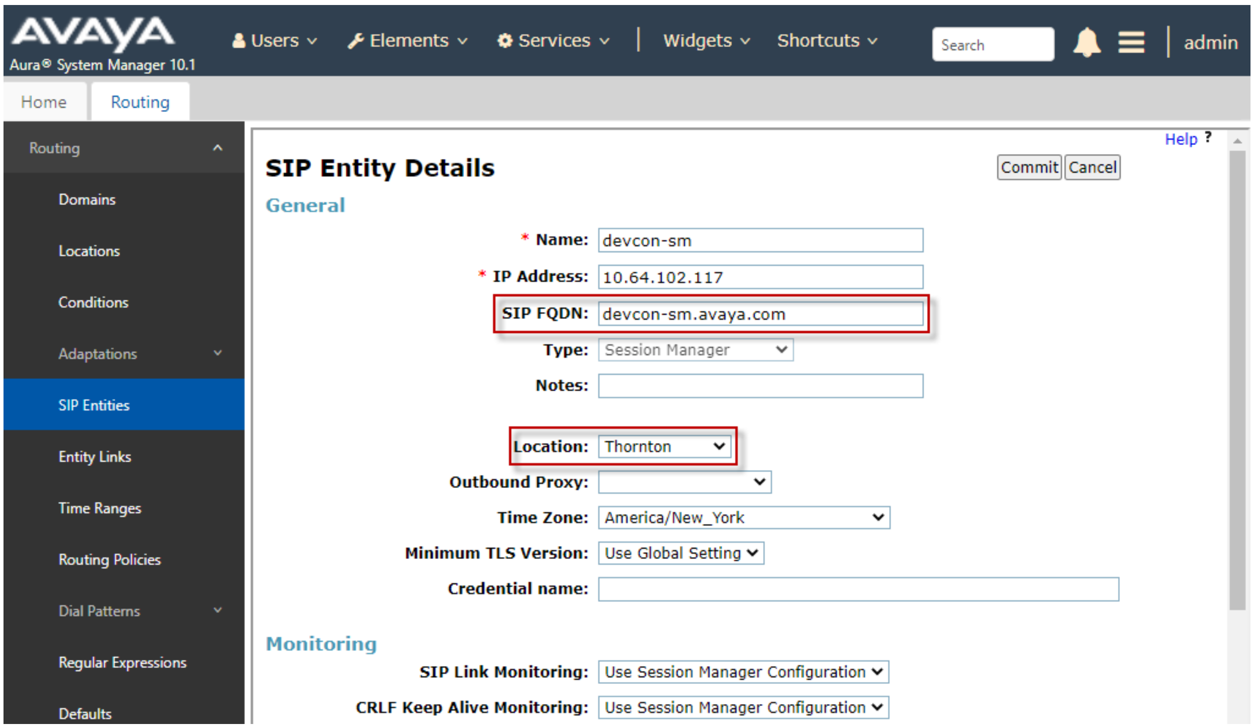
6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, navigate to **Elements → Routing → Locations**. The *Thornton* location was used for the DMP 128 Plus and all other SIP endpoints.

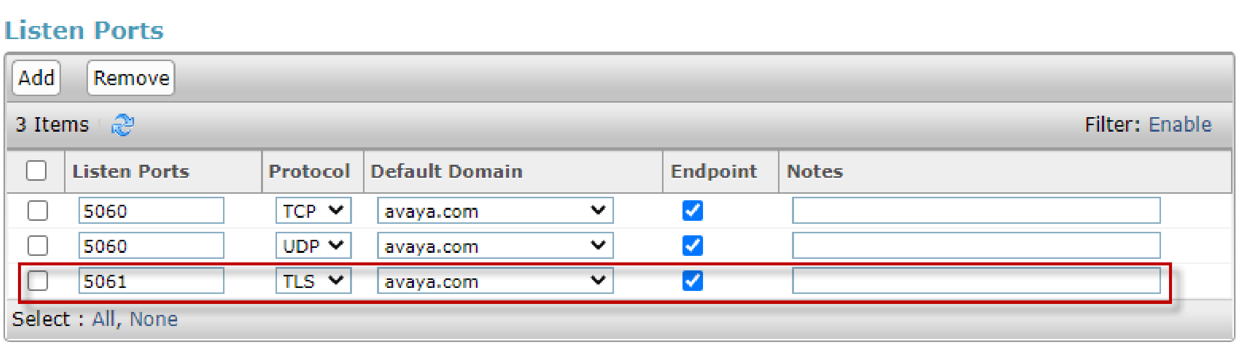


6.4. Administer Session Manager SIP Entity

From the System Manager **Home** screen, select **Elements → Routing → SIP Entities** and edit the SIP Entity for Session Manager shown below. Set **SIP FQDN** to the Session Manager FQDN determined in **Section 6.6**. This is required because DMP 128 Plus sends the FQDN in the Route header of the Register message and it needs to be resolved. In addition, specify the appropriate **Location** configured in **Section 6.3**.

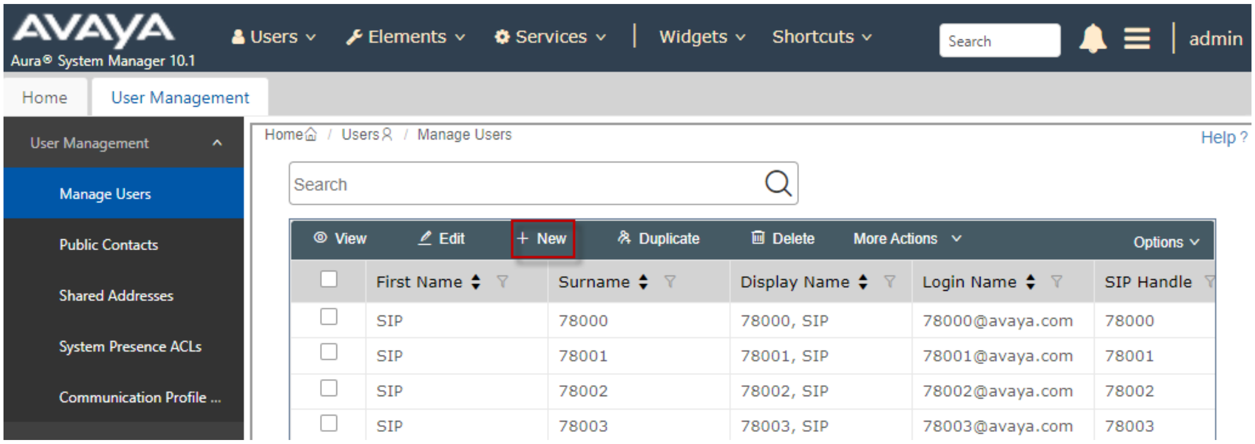


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by DMP 128 Plus is specified in the list below. For the compliance test, DMP 128 Plus used TLS transport. The **Default Domain** does not have to match the SIP domain used by DMP 128 Plus, because DMP 128 Plus sends it in SIP messages.



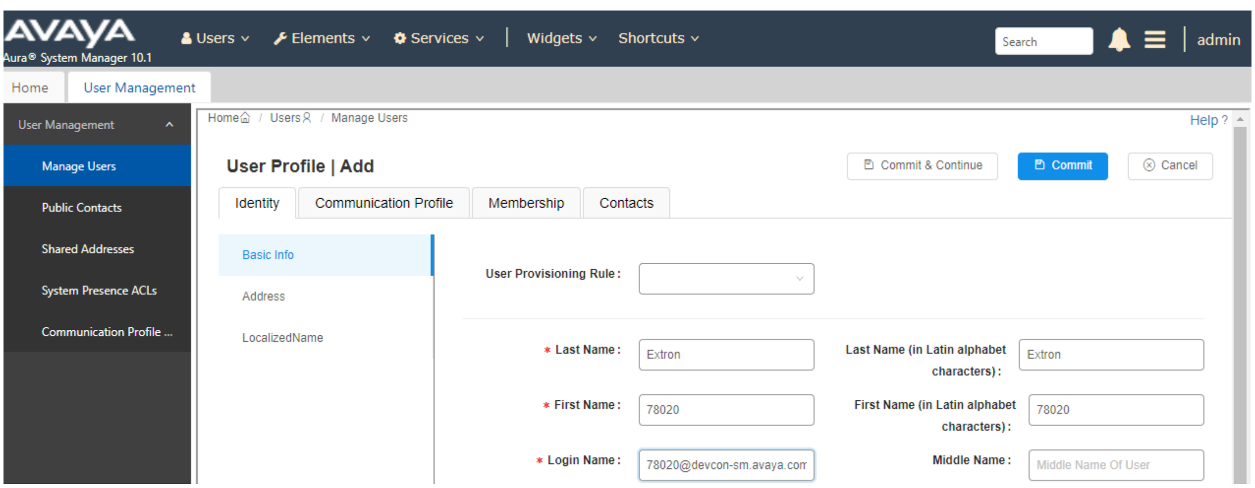
6.5. Administer SIP User

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.



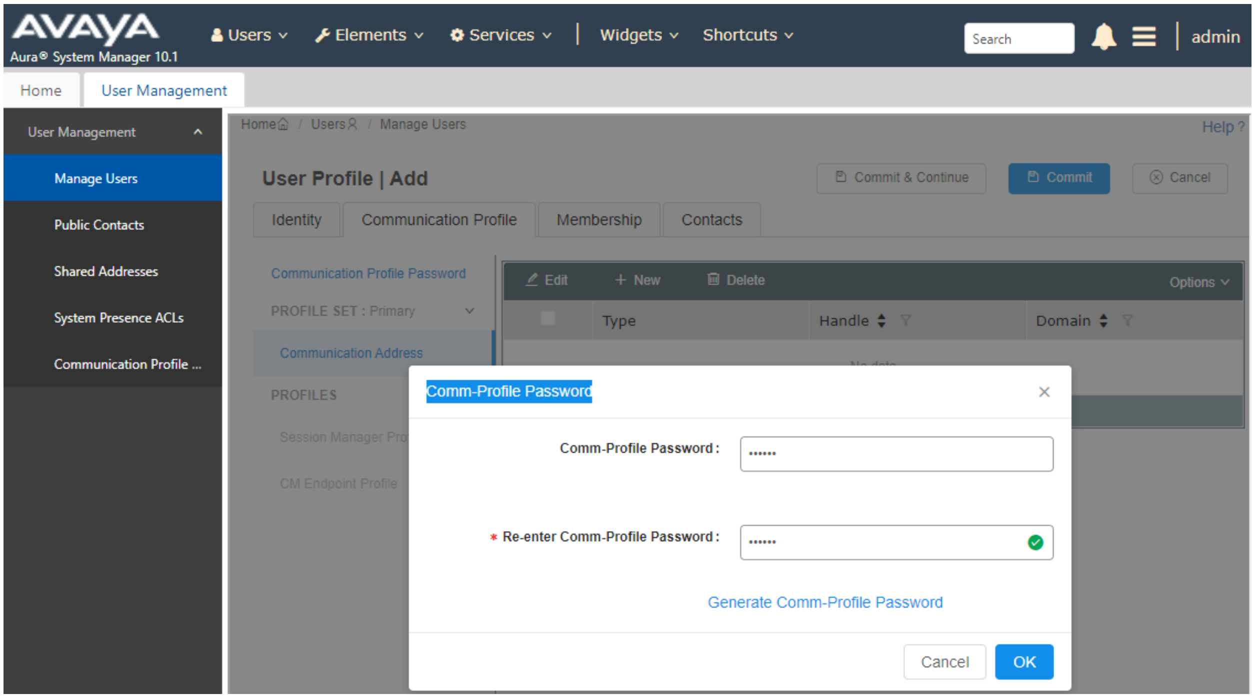
6.5.1. Identity

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.



6.5.2. Communication Profile

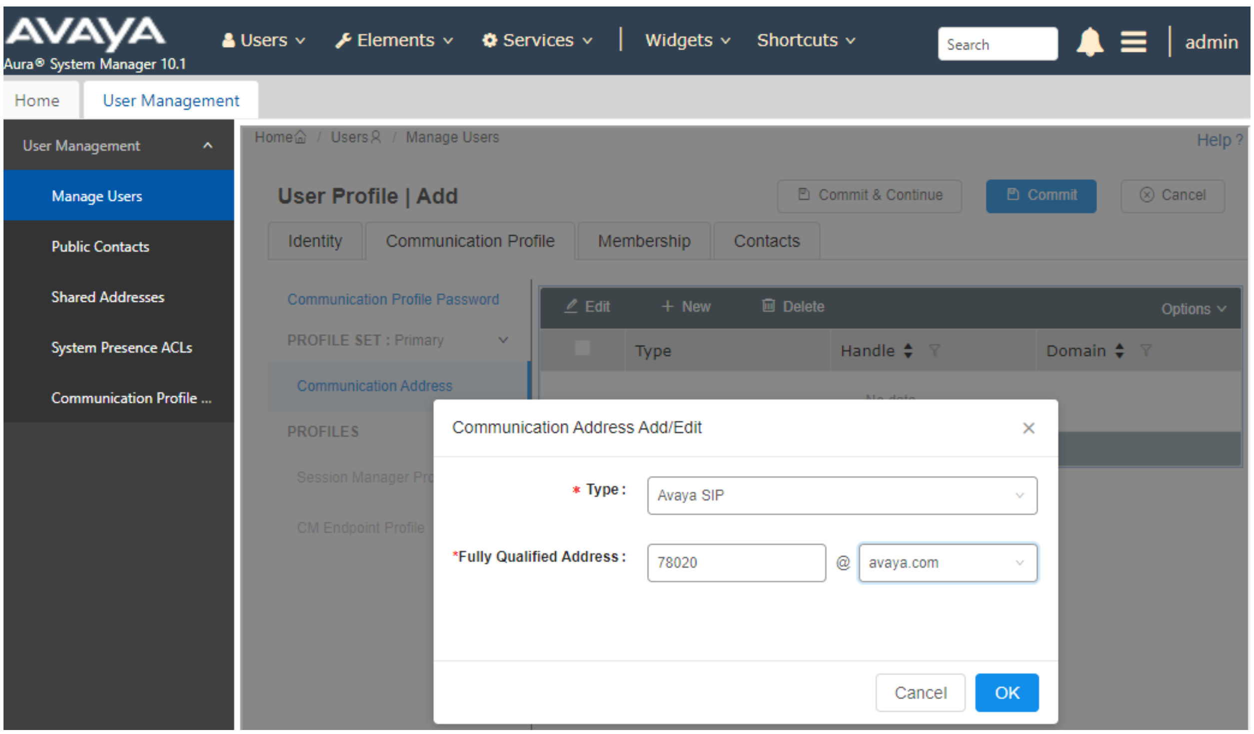
Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



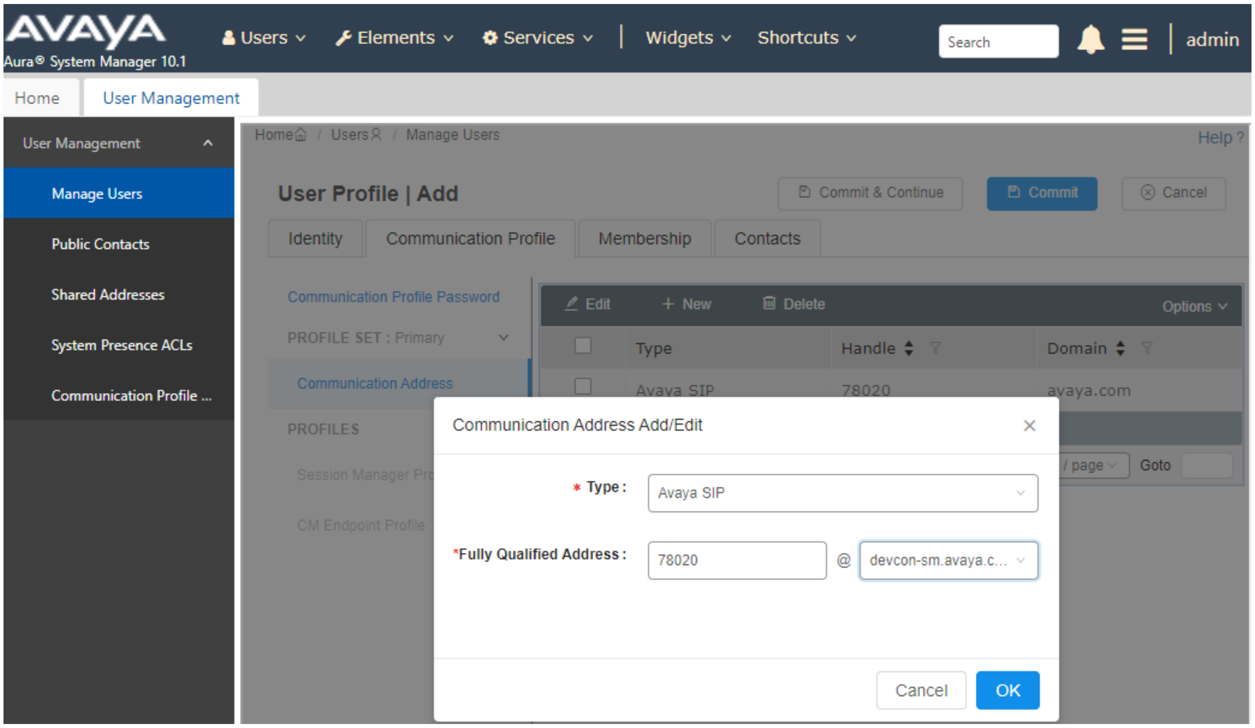
6.5.3. Communication Address

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 6.5.1**. Two Communication Addresses were added, one with the *avaya.com* domain name, which is used by existing SIP endpoints in calls to DMP 128 Plus, and one with the *devcon-sm.avaya.com* domain, which is used by DMP 128 Plus for registration and outgoing calls.

The following Communication Address specifies the *avaya.com* domain, which is used by existing SIP endpoints and Communication Manager when placing calls to DMP 128 Plus. If this domain is not included, the call would not be delivered to DMP 128 Plus.

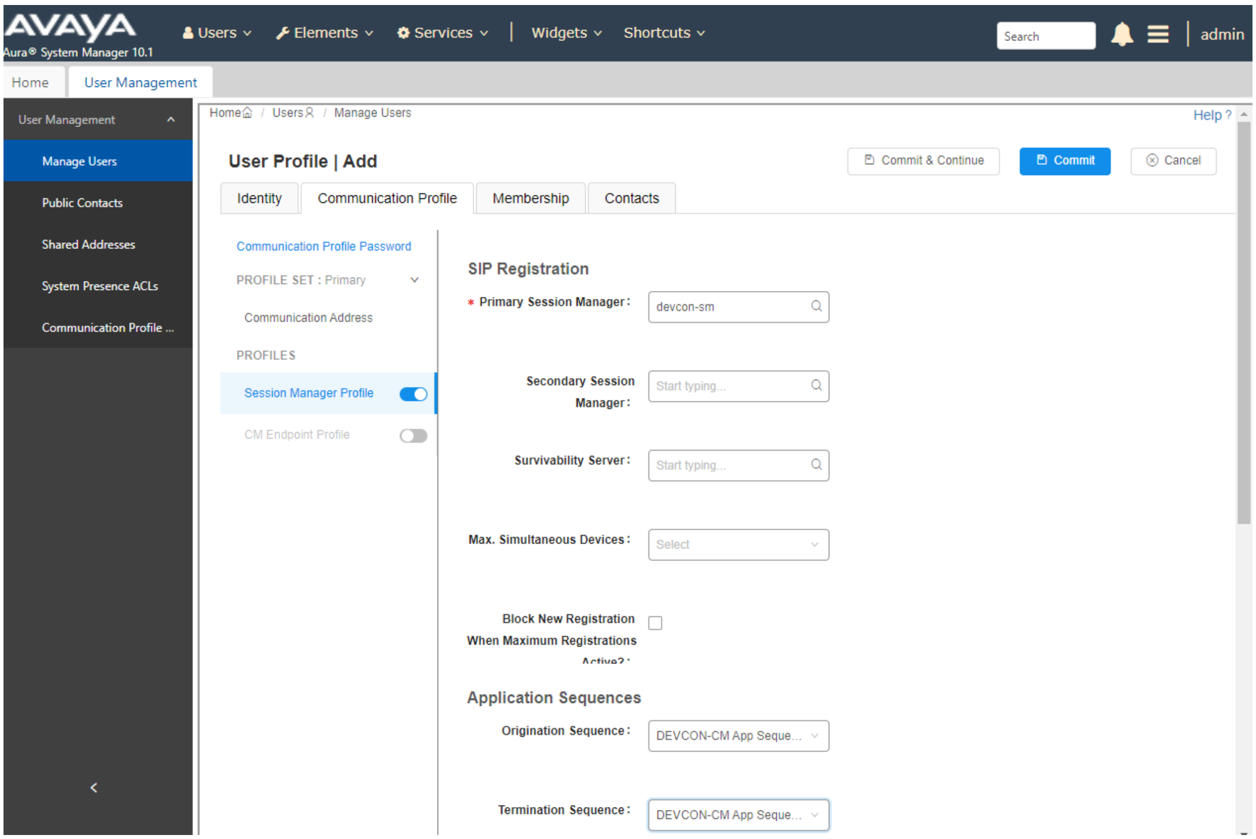


The following Communication Address specifies the *devcon-sm.avaya.com* domain, which is used by DMP 128 Plus during registration and outgoing calls. For outgoing calls from DMP 128 Plus, Communication Manager will change the domain to *avaya.com*. If this domain is not included, the DMP 128 Plus would not be able to register or place outgoing calls.

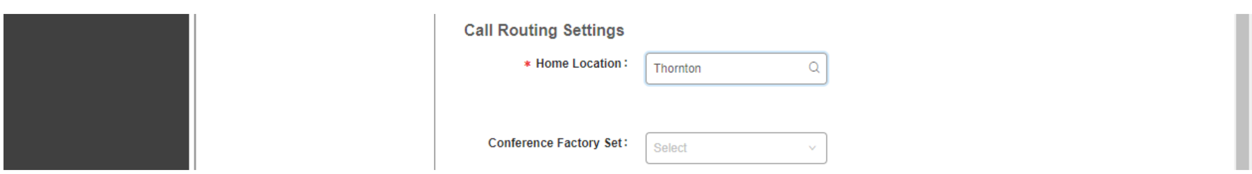


6.5.4. Session Manager Profile

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.

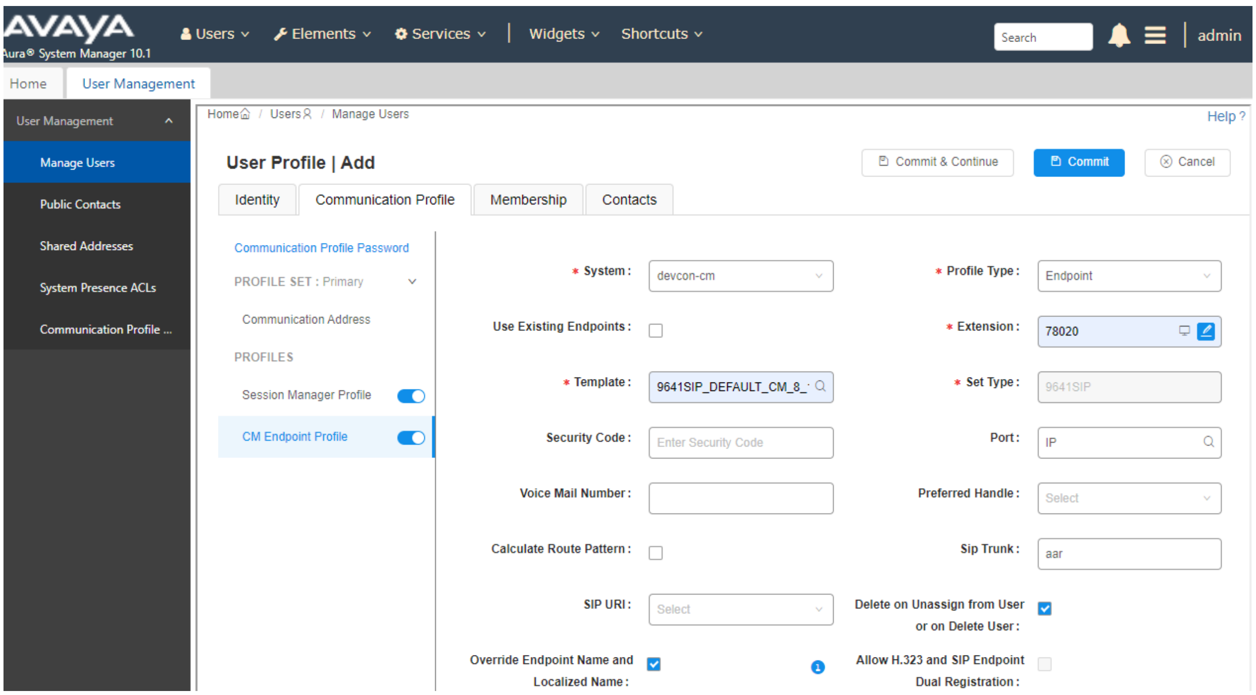


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

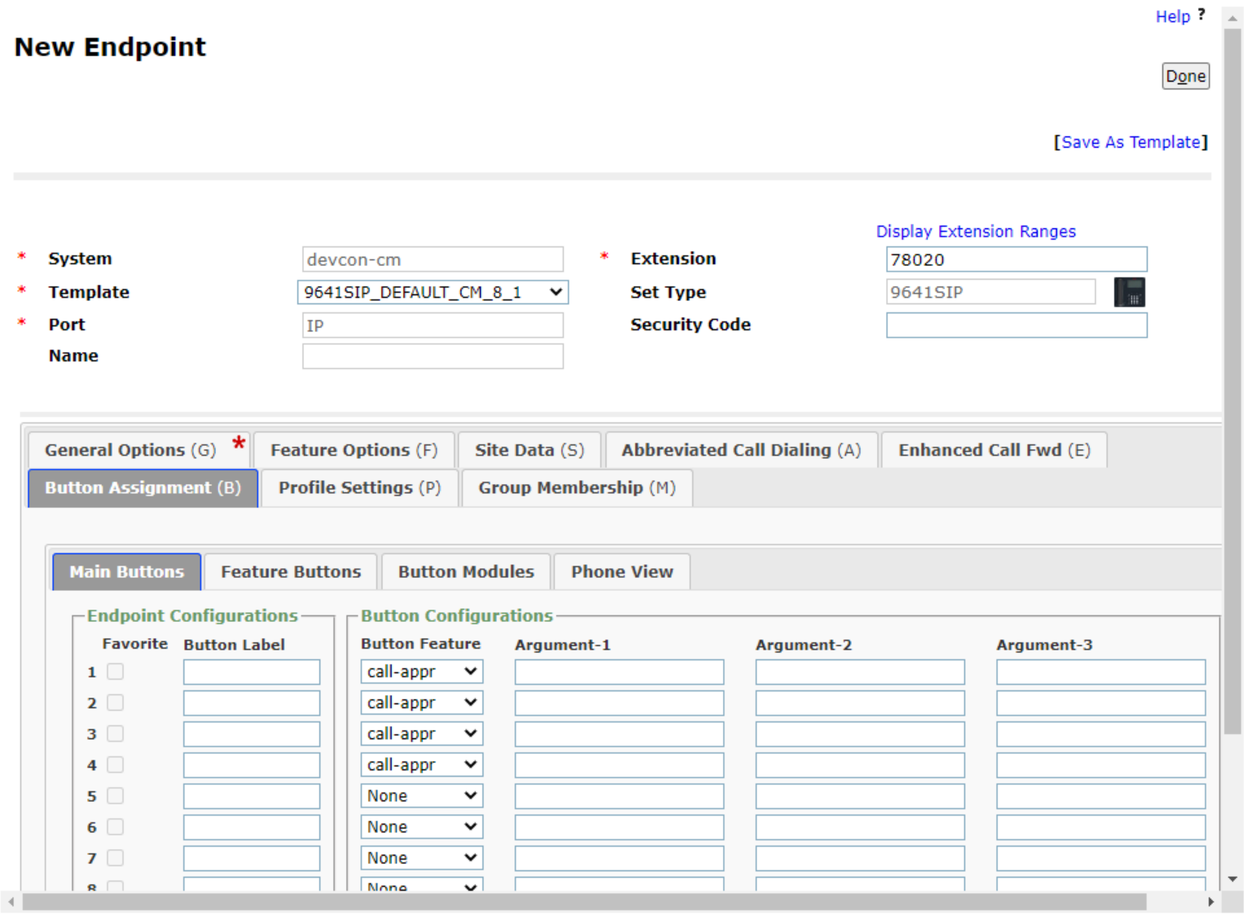


6.5.5. CM Endpoint Profile

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 6.5.3**. 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. Click **Done** to return to the previous web page and then **Commit** to save the configuration (not shown).



6.6. Verify Session Manager FQDN

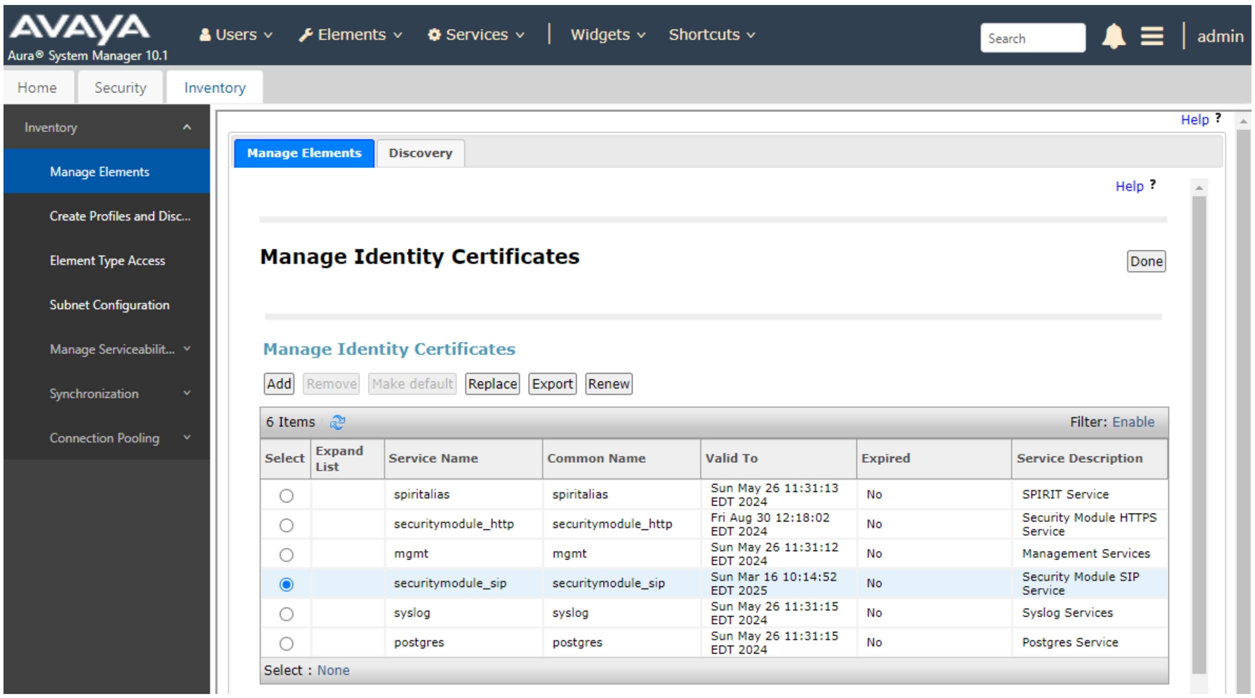
This section verifies the Session Manager FQDN, which is used during the DMP 128 Plus configuration, and also verifies that the Session Manager identity certificate includes it in the SAN.

Log into Session Manager via a SSH session using the appropriate credentials. Run the *smconfig* command, which displays various Session Manager parameters, including its FQDN as shown below.

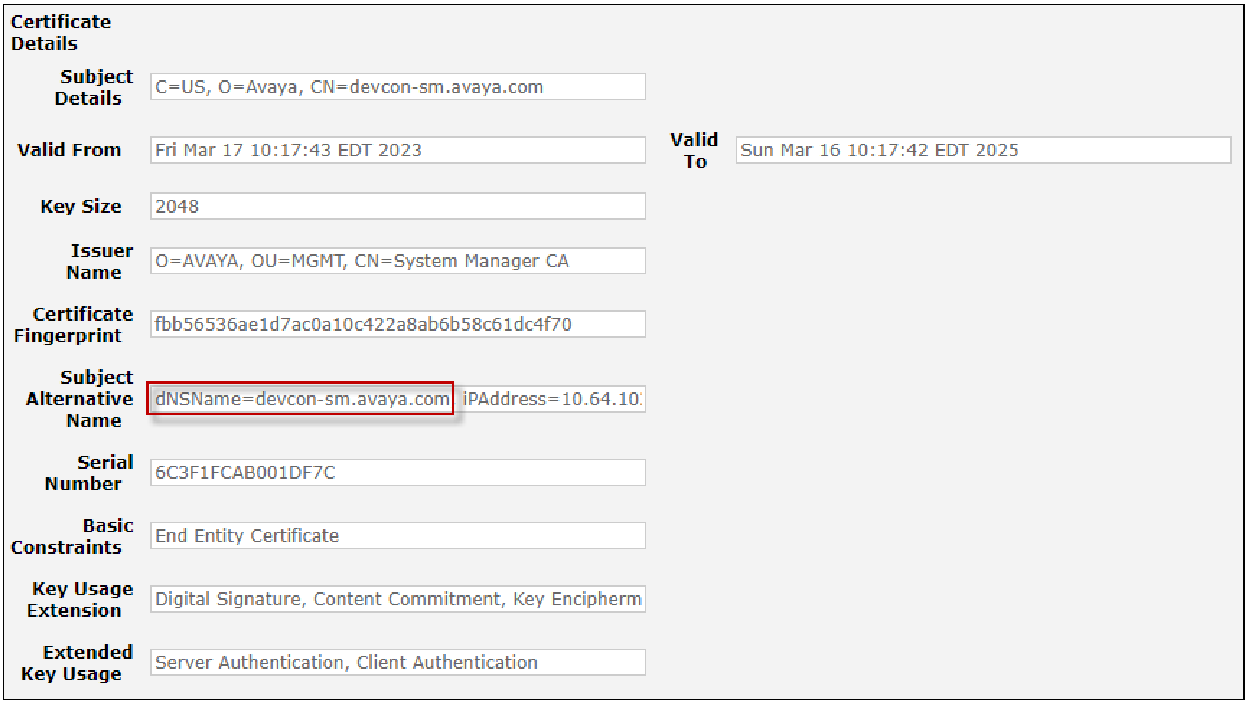
```
cust@devcon-sm:~
[cust@devcon-sm ~]$ smconfig
System Manager: 10.64.102.120 (devcon-smgr.avaya.com)

Certificates:
  SIP: CN = devcon-sm.avaya.com, O = Avaya, C = US
  HTTP: CN = 10.64.102.117, O = Avaya, C = US
  POSTGRES: CN = devcon-sm.avaya.com, O = Avaya, C = US
  SYSLOG: CN = devcon-sm.avaya.com, O = Avaya, C = US
  SPIRIT: C=US, O=Avaya, CN=devcon-sm.avaya.com
  MGMT: C=US, O=Avaya, CN=devcon-sm.avaya.com

Local Instance Name:   devcon-sm
Local Management Host: 10.64.102.116
Local Security Module: 10.64.102.117
Local SIP State:       Allow New SIP
Local FQDN:            devcon-sm.avaya.com
Local Netmask:         255.255.255.0
Local Gateway:         10.64.102.1
Local Timezone:        America/New_York (EDT, -0400)
Local Date/Time:       Fri Mar 17 11:10:38 EDT 2023
Local NTP Servers:     pool.ntp.org
Local DNS Servers:     75.75.75.75
Local Search List:     avaya.com
Deployment:            Virtual Machine
Hypervisor:            VMware
Data Encryption:       Disabled
SELinux:               Enforcing
Profile:               SM Profile 1 - 2000/2400 Devices
Security Policy:       Standard
OS Crypto Policy:      LEGACY
Log Retention:         30 Days

No other Session Managers provisioned!

[cust@devcon-sm ~]$
```

To verify the Session Manager identify certificate, navigate to **Services → Inventory → Manage Elements** and select the Session Manager element. Next, select **Manage Identity Certificates** from the **More Actions** drop-down field (not shown).

In the **Manage Identity Certificates** page, select the *securitymodule_sip* service as shown below. The Session Manager identity certificate is displayed below that screen.



In the **Certificate Details**, verify that the **Subject Alternate Name** (SAN) contains the Session Manager FQDN as shown below.



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:

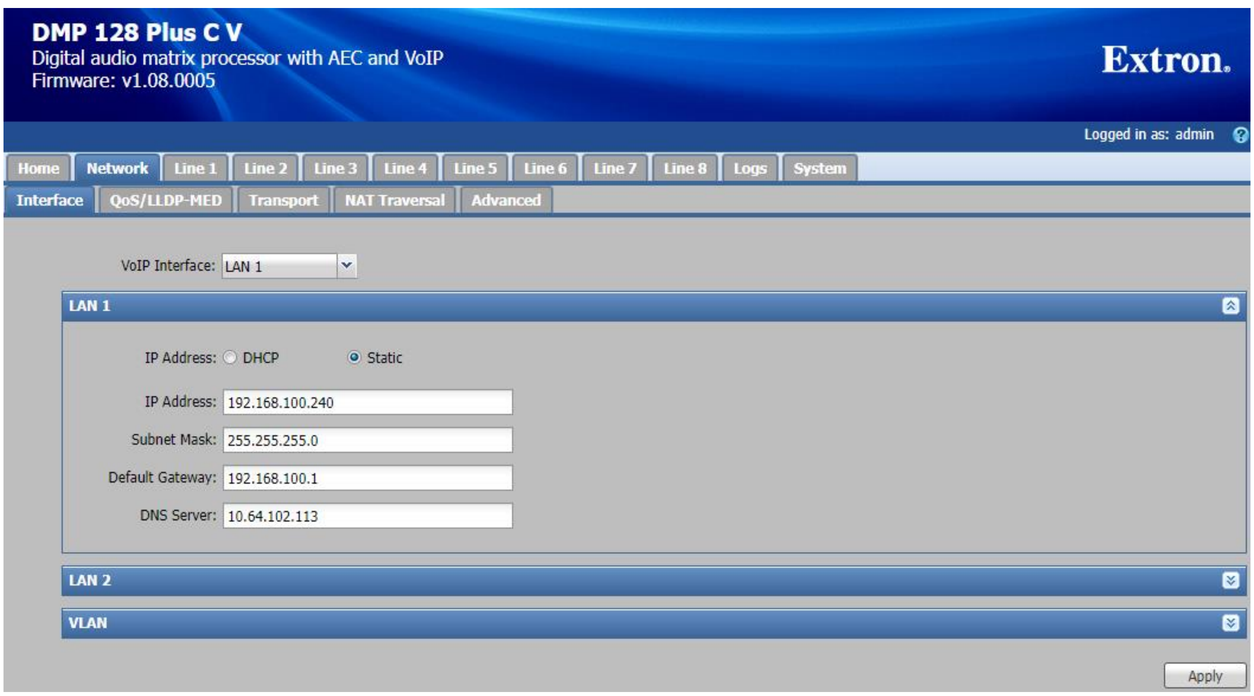
- Launch Web Interface
- Administer Network Settings
- Administer SIP Settings
- Configure the DSP

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 DMP 128 web interface is displayed as shown in the following.

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, *192.168.100.240*, was assigned to DMP 128 Plus. In addition, **DNS Server** must be configured so that the Session Manager FQDN, configured as the **Primary Proxy Name** of **Line 1** in **Section 7.3**, could be resolved. Alternatively, DHCP may be used. Click **Apply**.

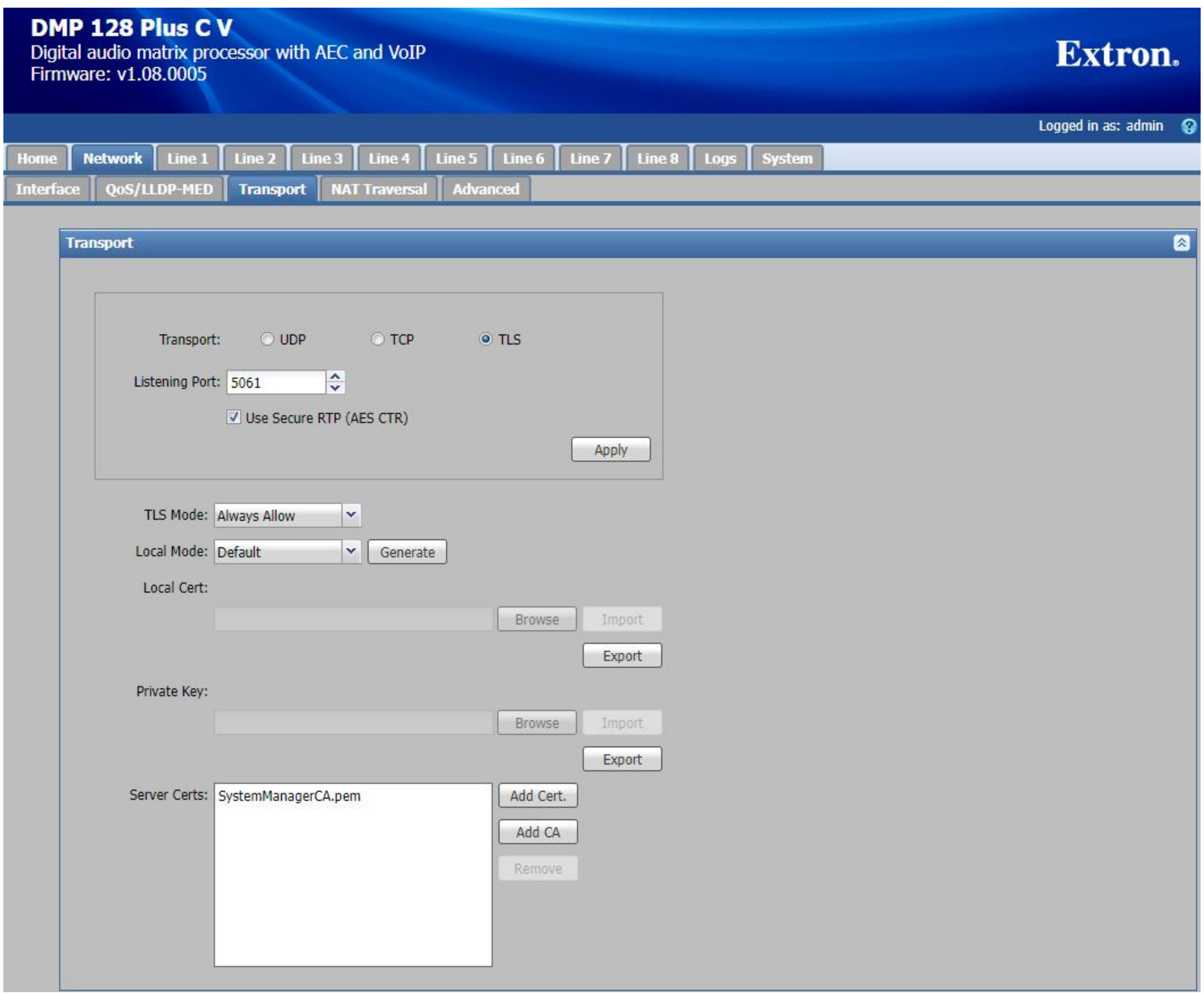


7.3. Administer SIP Settings

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 *TLS* transport protocol.
- **Listening Port:** Specify port *5061*.
- **Use Secure RTP (AES CTR):** Enable SRTP.
- **TLS Mode:** Set to *Always Allow* or *Always Verify*. If *Always Verify* is used, the Session Manager hostname will be verified during the TLS handshake. This requires that the Session Manager FQDN is configured in the SAN of the Session Manager identity certificate as shown in **Section 6.6**.

Click **Apply**. In the **Server Certs** section, import the appropriate CA certificate. For this compliance test, the System Manager CA certificate was used.

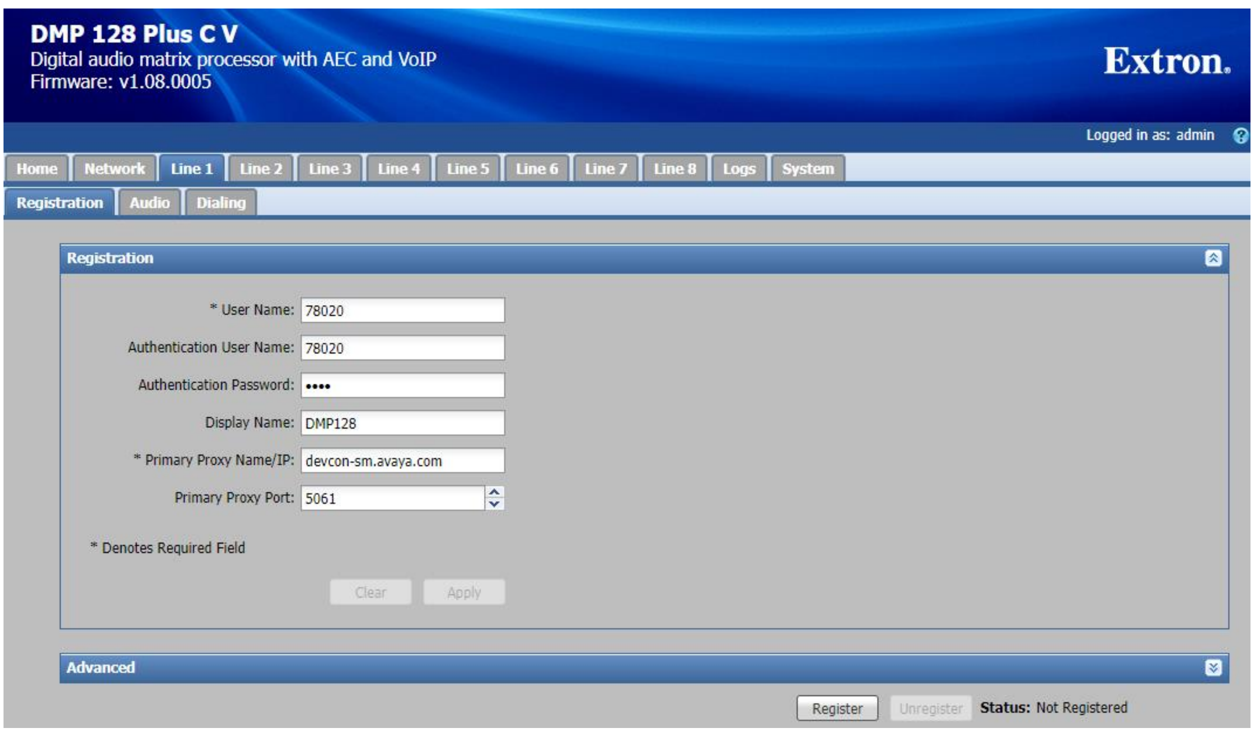


Navigate to **Line 1** tab to configure the SIP registration settings. DMP 128 Plus uses the **Server Name Indication** (SNI) extension during the TLS handshake with Session Manager. The **Primary Proxy Name/IP** field must be set to the Session Manager FQDN (e.g., *devcon-sm.avaya.com*), not the Session Manager IP address. As a result, DMP 128 Plus also uses the SNI as the SIP domain for SIP registration and outgoing calls. However, DMP 128 Plus will also accept incoming calls using the *avaya.com* domain, which was used by other SIP endpoints in the compliance test.

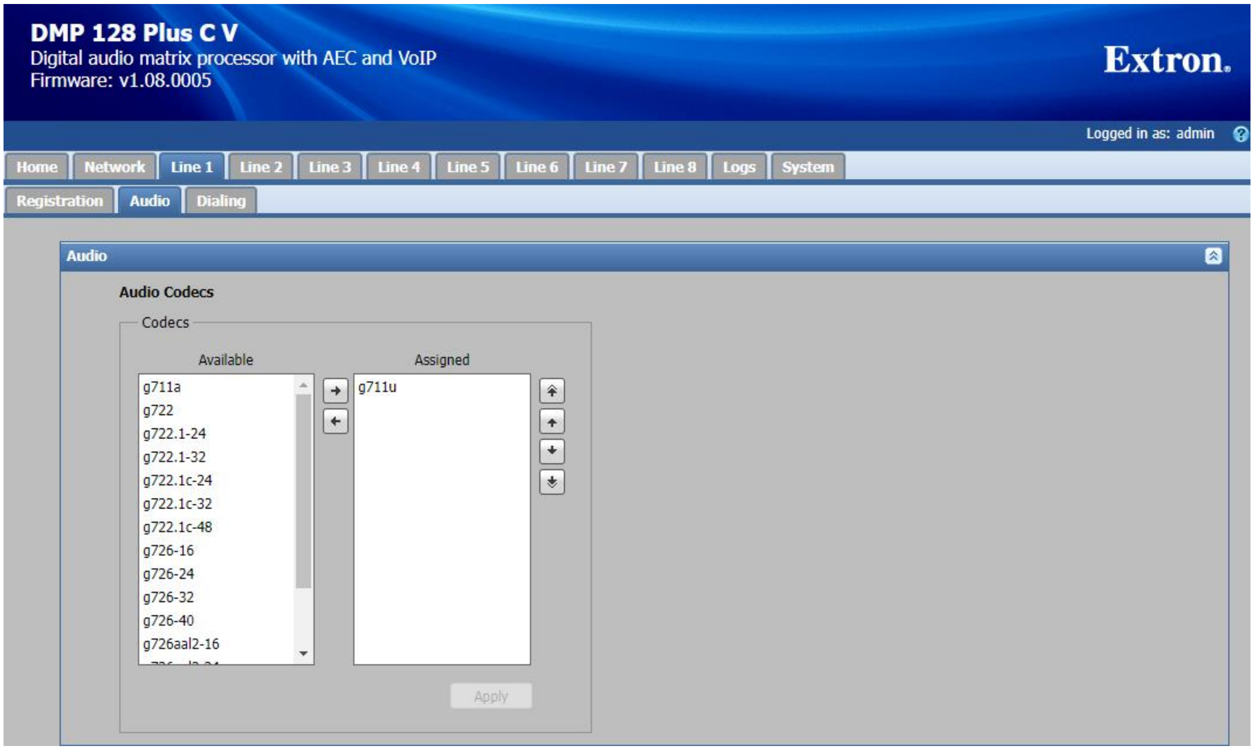
Configure the following additional fields:

- **User Name** Specify the SIP extension (e.g., *78020*).
- **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 FQDN (e.g., *devcon-sm.avaya.com*) as determined in **Section 6.6**.
- **Primary Proxy Port** Specify SIP port *5061* .

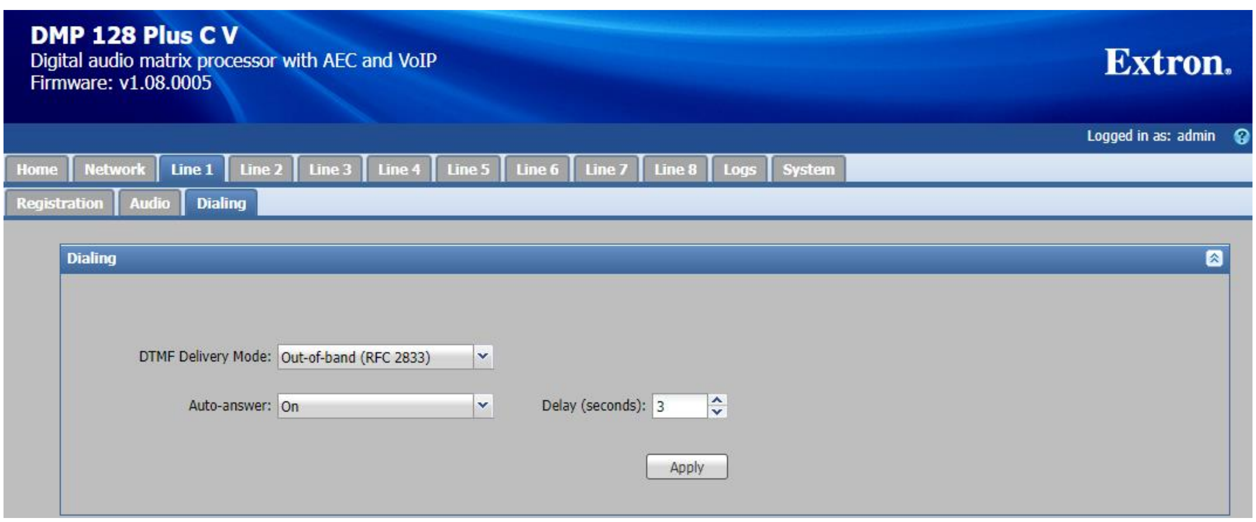
Click **Apply** to submit the changes and then click the **Register** button to register DMP 128 Plus with Session Manager after all the configuration is in place.



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



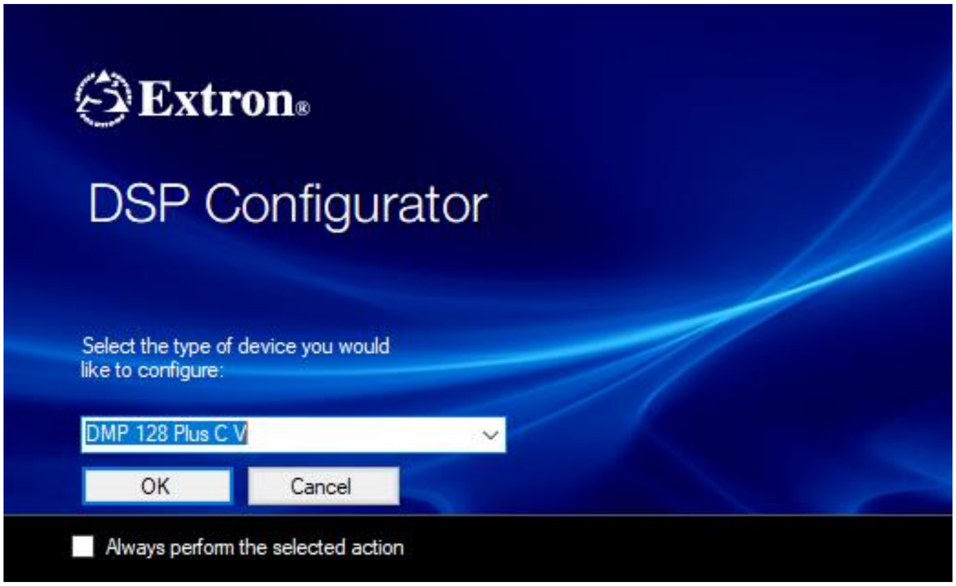
In the **Dialing** sub-tab, accept the default settings shown below. The **DTMF Delivery Mode** is set to *Out-of-band (RFC 2833).*



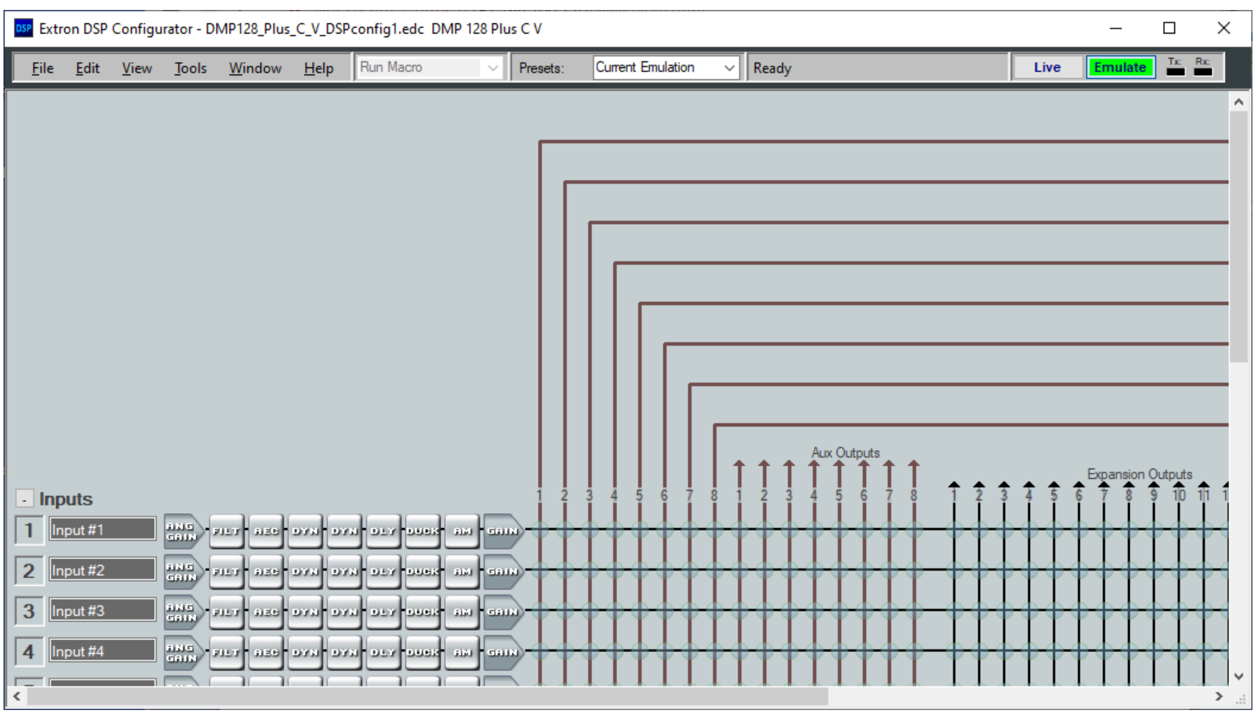
7.4. Configure the DSP

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

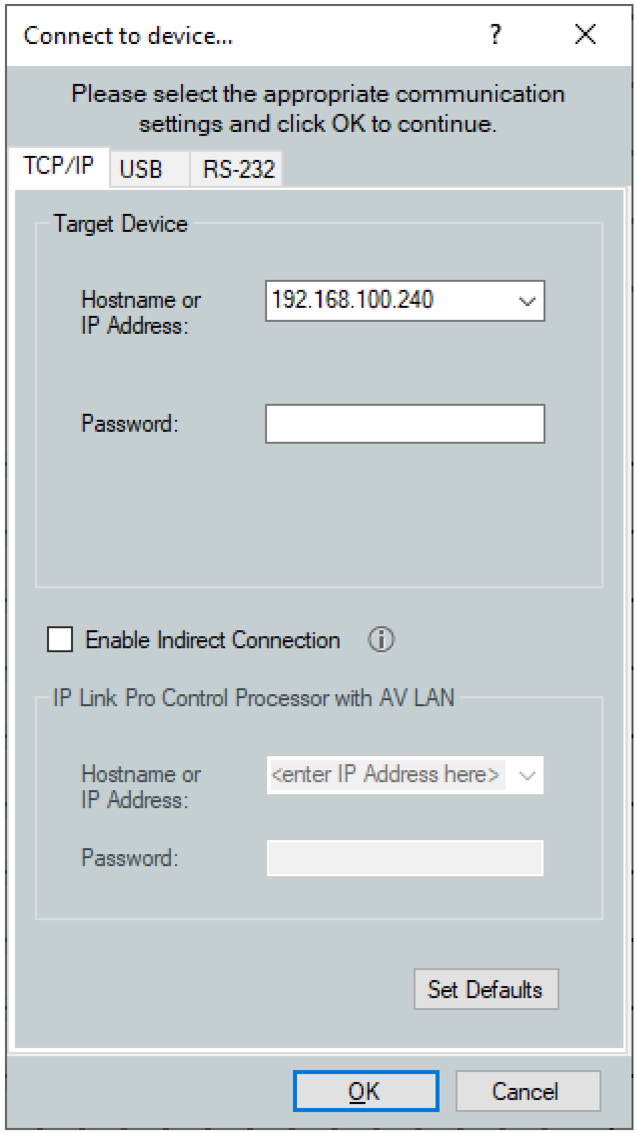
Launch the **DSP Configurator**, select the device type in the drop-down field and click **OK**.



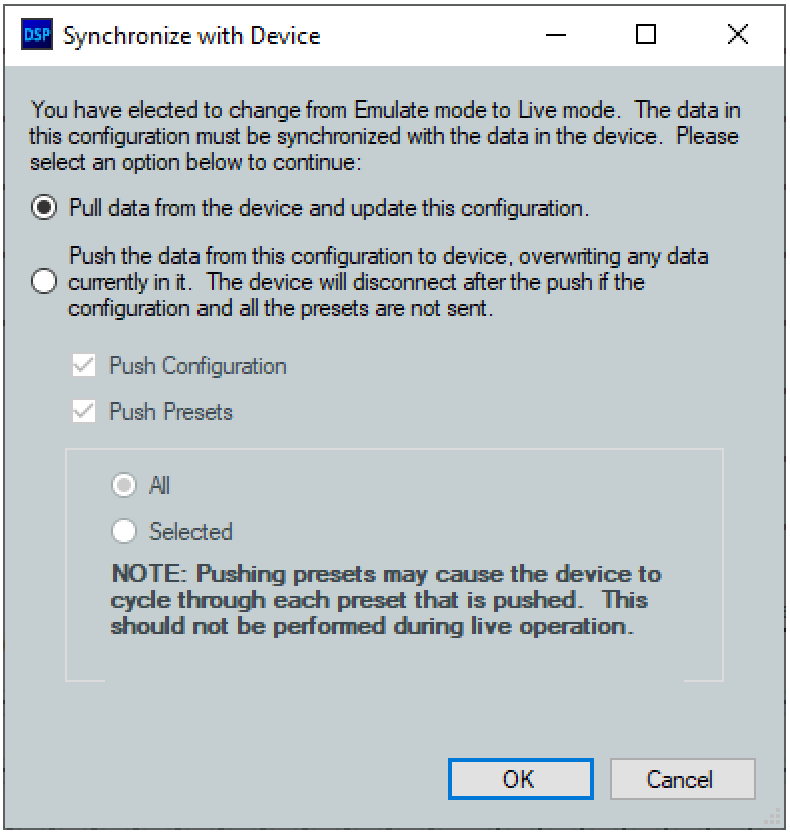
The following screen is displayed. Click the Live button to connect to the device.



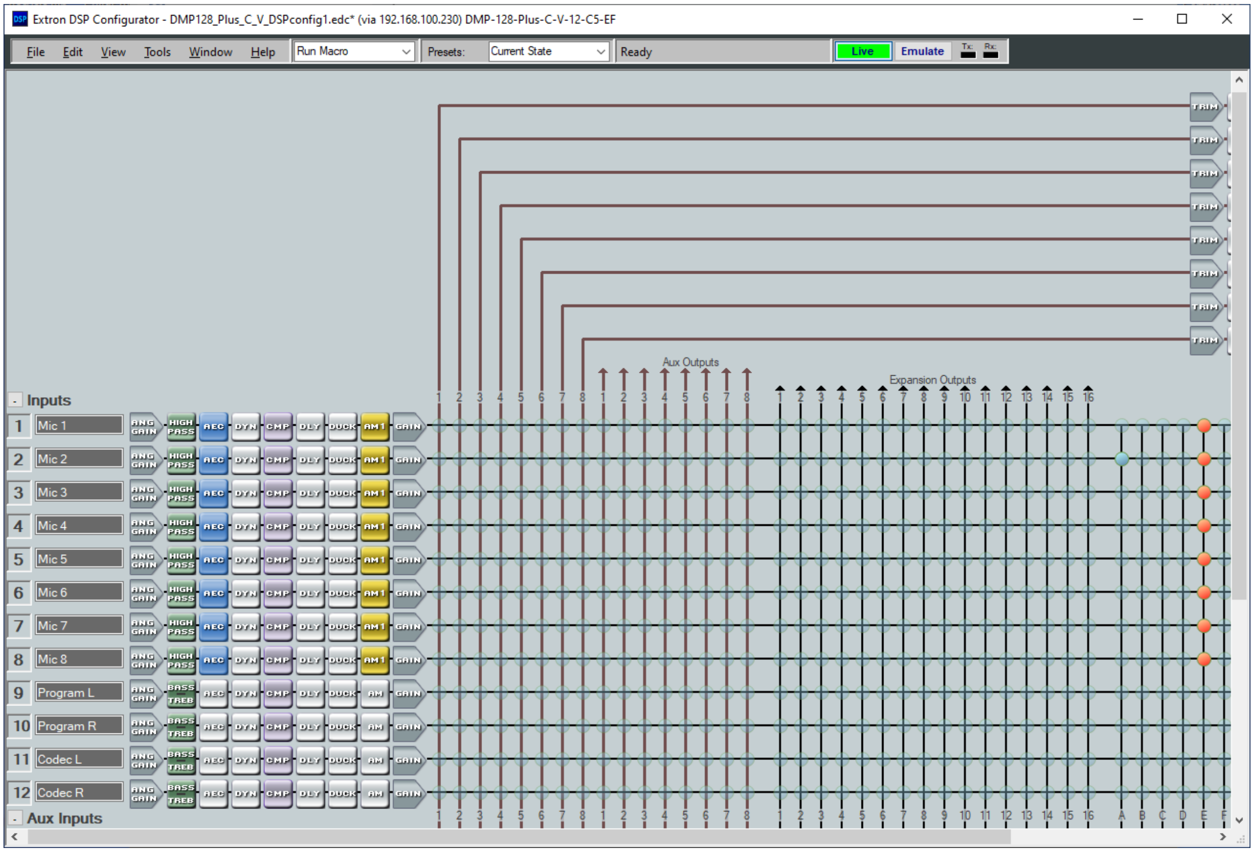
In the **Connect to device…** window shown below, enter the DMP 128 Plus IP address in the **Hostname or IP Address** field and click **OK**.



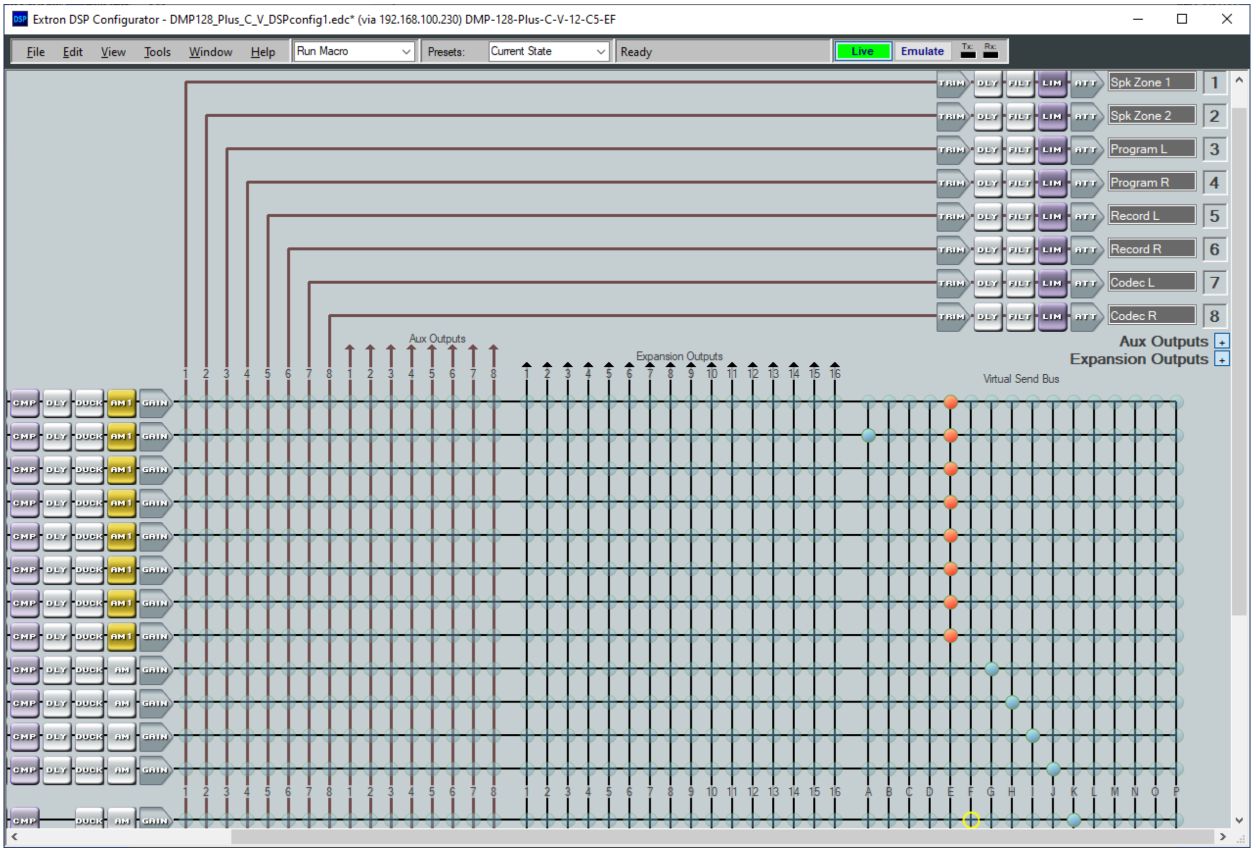
In the **Syncrohonize with Device** window shown below, select *Pull data from the device and update the configuration* and click **OK**.



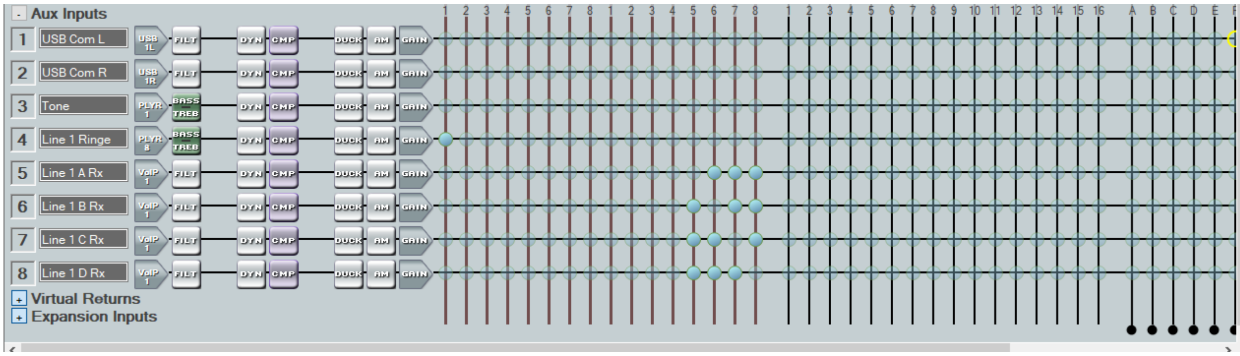
Once connected to DMP 128 Plus, the **DSP Configurator** is displayed as shown below. The following displays the top left portion of the screen.



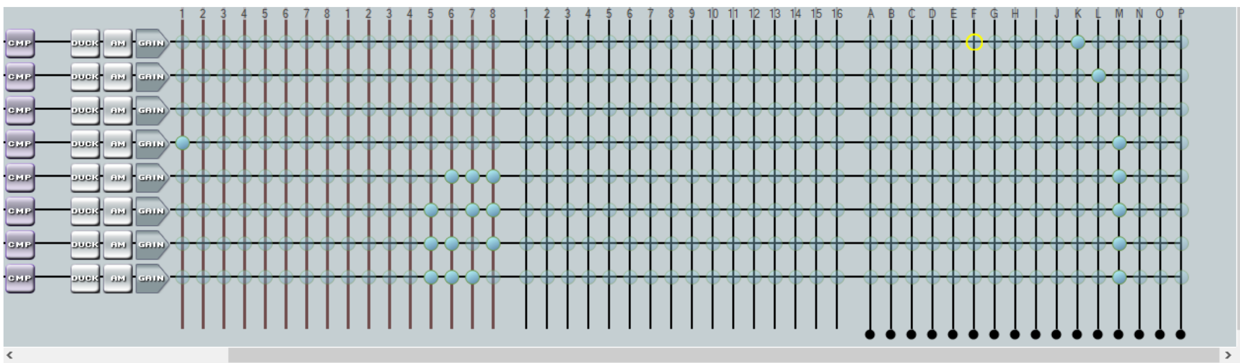
Scroll to the right to view the rest of the screen.



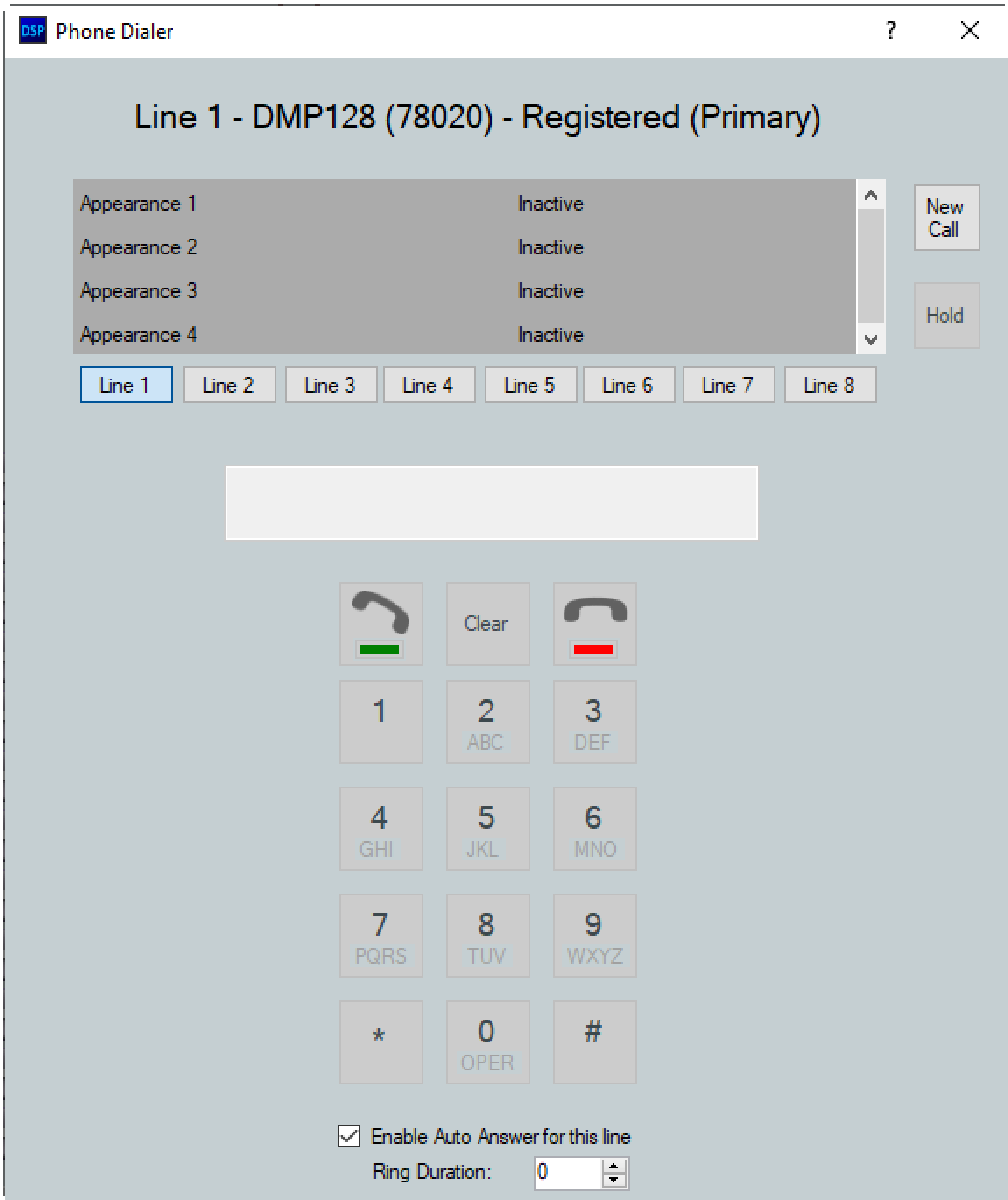
Scrolling down shows that four call appearances were configured for the SIP line and that the audio call appearances were mixed together. This results in all calls being conferenced together automatically.



Scroll to the right to view 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 TouchLink Pro**, a customizable touch panel, would be used by customers for this purpose, which would provide audio tone feedback for each call.



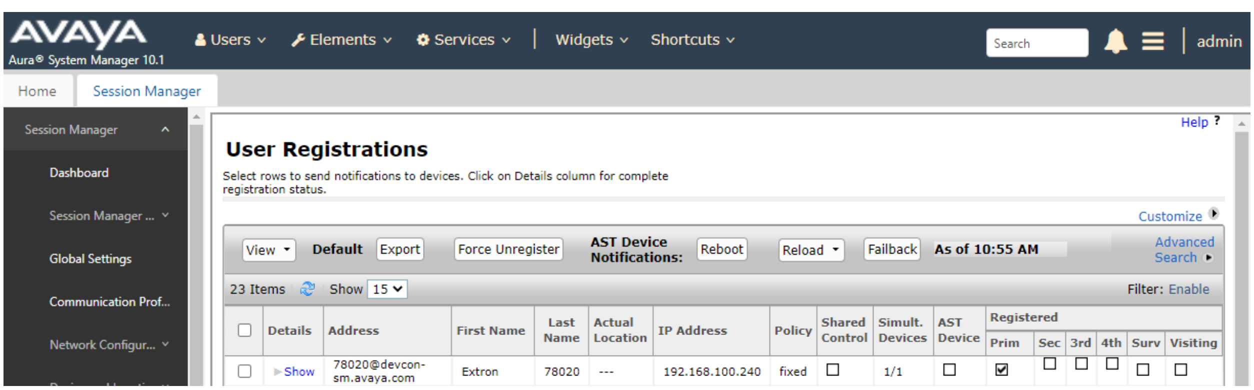
8. Verification Steps

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

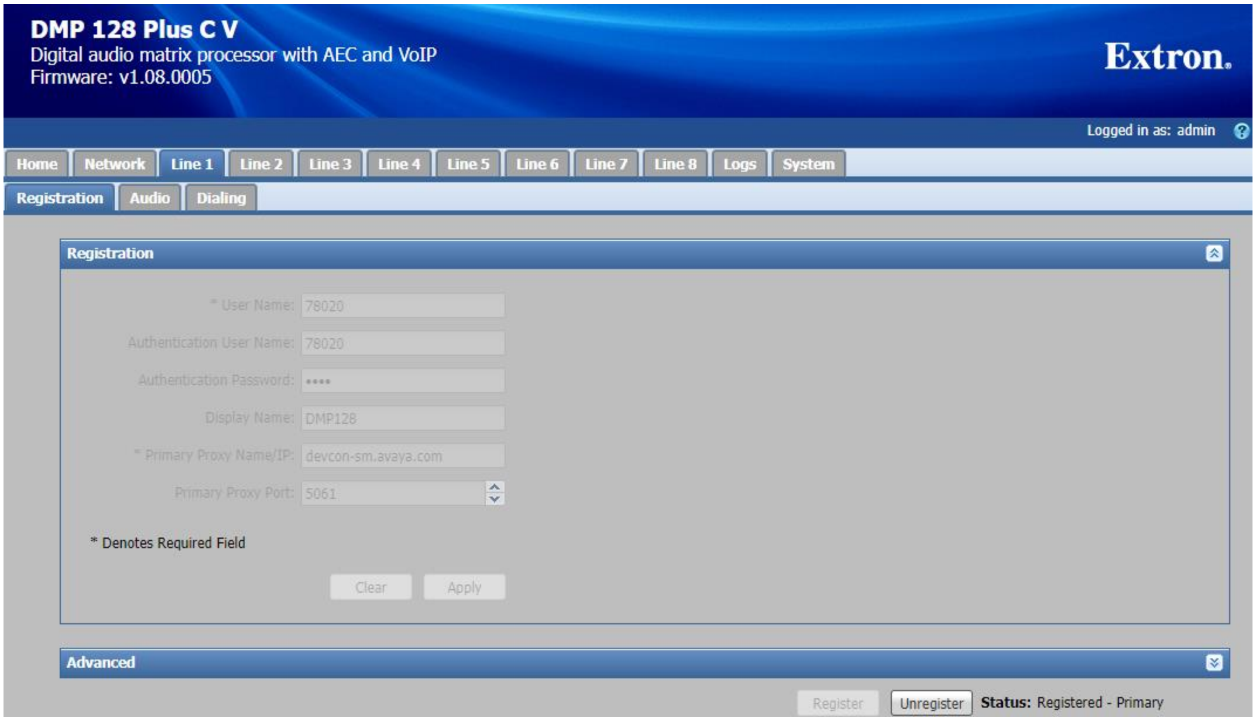
1. During the TLS handshake, verify that DMP 128 Plus sends the Session Manager server name in the Server Name Indication extension of the TLS Client Hello message as shown below.

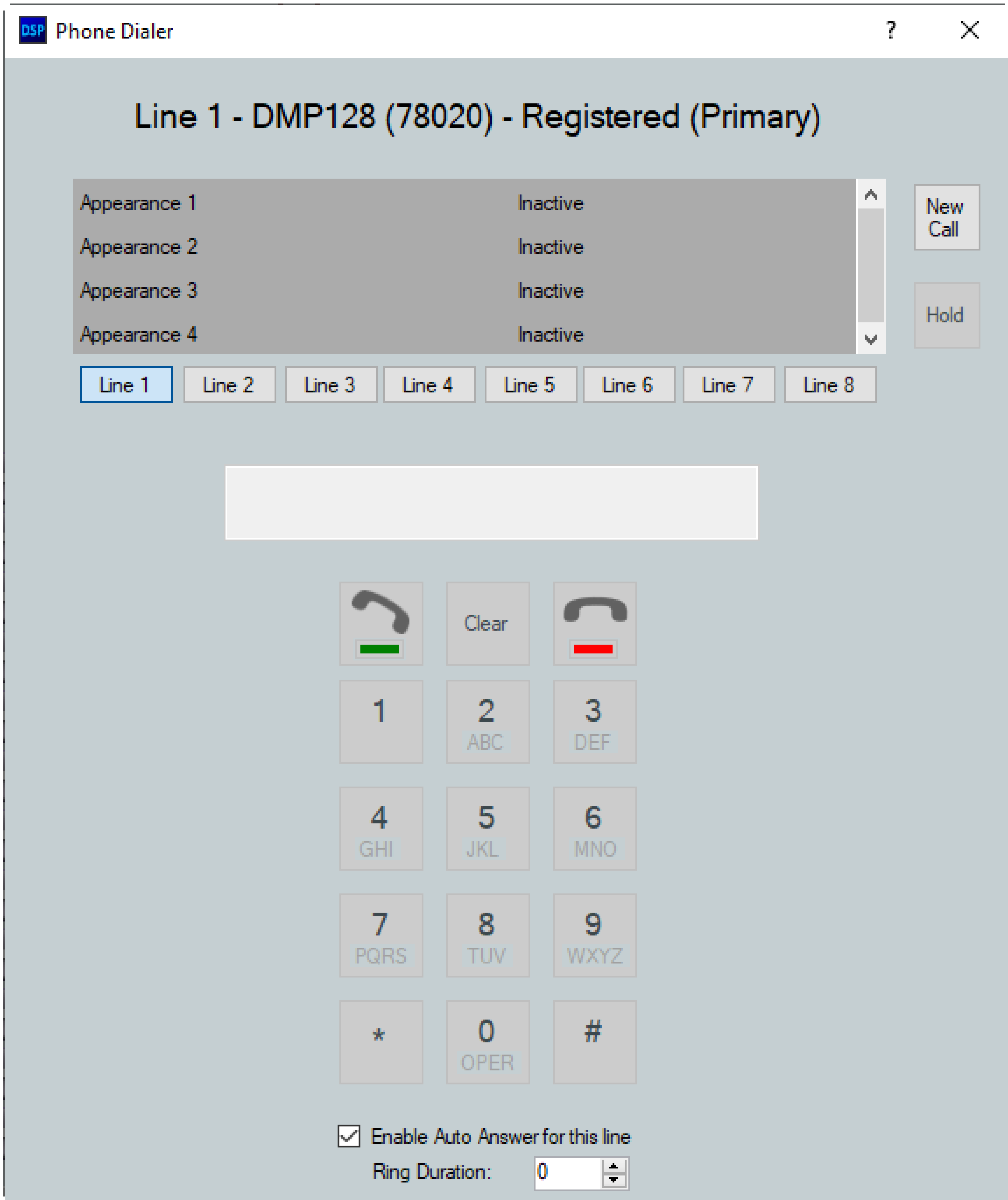
```
    Cipher Suite: TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA256 (0xc025)
    Cipher Suite: TLS_ECDH_RSA_WITH_AES_128_CBC_SHA (0xc00e)
    Cipher Suite: TLS_ECDH_ECDSA_WITH_AES_128_CBC_SHA (0xc004)
    Cipher Suite: TLS_RSA_WITH_AES_128_GCM_SHA256 (0x009c)
    Cipher Suite: TLS_RSA_WITH_AES_128_CBC_SHA256 (0x003c)
    Cipher Suite: TLS_RSA_WITH_AES_128_CBC_SHA (0x002f)
    Cipher Suite: TLS_RSA_WITH_CAMELLIA_128_CBC_SHA (0x0041)
    Cipher Suite: TLS_EMPTY_RENEGOTIATION_INFO_SCSV (0x00ff)
Compression Methods (2 methods)
    Compression Method: DEFLATE (1)
    Compression Method: null (0)
Extension: server_name (len=24)
    Server Name Indication extension
        Server Name list length: 22
        Server Name Type: host_name (0)
        Server Name length: 19
        Server Name: devcon-sm.avaya.com
Extension: ec_point_formats (len=4)
    Elliptic curves point formats (3)
        EC point format: uncompressed (0)
        EC point format: ansiX962_compressed_prime (1)
        EC point format: ansiX962_compressed_char2 (2)
Extension: supported_groups (len=28)
    Supported Groups List Length: 26
    Supported Groups (13 groups)
        Supported Group: secp256r1 (0x0017)
        Supported Group: secp521r1 (0x0019)
```

2. 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.



3. 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.





4. 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 10.1.x, Issue 4, February 2023.
[2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 8, February 2023.
[3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 5, February 2023.
[4] *Extron DMP 128 Plus User Guide*, 68-2826-01 Rev. K, 11 22.

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.

ATTACHMENT 1

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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