Avaya Solution & Interoperability Test Lab

Application Notes for Symetrix 2 Line VoIP Interface Card with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Symetrix 2 Line VoIP Interface Card with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The 2 Line VoIP Interface Card supports a range of telephony functions, including dial, hold, resume, transfer, and conference, and registers with Avaya Aura® Session Manager as a SIP endpoint. The 2 Line VoIP Interface Card is a SIP-based plug-in card for Symetrix Radius AEC and Radius Edge products. For this compliance test, the 2 Line VoIP Interface Card was installed in Symetrix Radius AEC. Radius AEC used the 2 Line VoIP Interface Card for audio signaling processing for conferencing and sound reinforcement in distance learning and meeting (i.e., conference room) applications.

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 the Symetrix 2 Line VoIP Interface Card with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The 2 Line VoIP Interface Card supports a range of telephony functions, including dial, hold, resume, transfer, and conference, and registers with Avaya Aura® Session Manager as a SIP endpoint. The 2 Line VoIP Interface Card is a SIP-based plug-in card for Symetrix Radius AEC and Radius Edge products. For this compliance test, the 2 Line VoIP Interface Card was installed in Symetrix Radius AEC. Radius AEC used the 2 Line VoIP Interface Card for audio signaling processing for conferencing and sound reinforcement in distance learning and meeting (i.e., conference room) applications.

With the 2 Line VoIP Interface Card, Radius AEC was able to establish or participate in an audio conference with parties on local stations or PSTN via the Avaya SIP-based network. Other participants in a meeting room or class room, where Radius AEC is located, could then communicate with the conference participants via a microphone and speakerphone connected to Radius AEC.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Symetrix 2 Line VoIP Interface Card (installed in Symetrix Radius AEC), Avaya SIP and H.323 IP Deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer and conference. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs) and Feature Name Extensions (FNEs).

The serviceability testing focused on verifying that the Symetrix 2 Line VoIP Interface Card came back into service after re-connecting the Ethernet cable or rebooting the Symetrix Radius AEC.

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.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of 2 Line VoIP Interface Card with Session Manager.
- Calls between Radius AEC with 2 Line VoIP Card and Avaya SIP/H.323 IP Deskphones with Direct IP Media (Shuffling) enabled and disabled.

- Calls between 2 Line VoIP Interface Card and the PSTN.
- G.711, G.729 and G.722 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, transfer, and 3-party conference.
- Extended telephony features using Communication Manager FACs and FNEs for Call Forward, Call Park/Unpark, and Call Pickup.
- Use of programmable buttons on 2 Line VoIP Interface Card.
- Proper system recovery after a restart of Radius AEC with 2 Line VoIP Interface Card and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation(s):

- Blind conference is not supported, but attended/supervised conference is supported.

- When supporting a blind transfer (i.e., completing a transfer prior to the transfer-party answering the call or while it is ringing), Symetrix 2 Line VoIP Interface Card doesn't drop off the call until the transfer-to party answers the call. To initiate a blind transfer, the VoIP card establishes a call and then initiates the transfer by pressing the transfer button the first time. The transfer-to party is then dialed and starts ringing. Prior to the transfer-to party being answered, the VoIP card completes the transfer by pressing the transfer button a second time. The VoIP card doesn't drop off the call until the transfer-party answers the call at which time the call is transferred immediately and the VoIP card drops off the call. Typically, with a blind transfer, the party (in this case the VoIP card) initiating the transfer drops off the call before the transfer-to party answers the call.

- When Radius AEC with the 2 Line VoIP Interface Card places a SIP call on hold, the 2 Line VoIP Interface Card sends a SIP INVITE with a "epv" parameter in the Request URI. The "epv" parameter contains a syntax error resulting in Communication Manager ignoring the SIP message. The VoIP card re-sends the SIP INVITE message a total of 9 times, and then eventually, the call fails (i.e., the call is dropped). The VoIP card includes the "epv" parameter in the SIP INVITE after pressing the hold button, because the Endpoint-View header was received from Session Manager during the establishment of the call. The Endpoint-View header is not required by the VoIP card to process calls. If Session Manager does not send the Endpoint-View header to the VoIP card, then the VoIP card won't include it in the SIP INVITE that is sent after pressing the hold button.

    - The **workaround** is to remove the Endpoint-View header from the SIP INVITE before it is sent to the VoIP Card using an Adaptation configured on Session Manager (see **Section 6.1**). However, this adaptation should only be applied to calls involving the VoIP card. This adaptation should not affect any other call.

      As part of this workaround, a separate SIP trunk between Communication Manager and Session Manager is configured (see **Section 5.5.1**) and dedicated for calls to the VoIP card. All other calls will use a different SIP trunk (see **Section 5.5.2**). The SIP trunk dedicated to calls to the VoIP card will be associated with a SIP entity on Session Manager (see **Section 6.2.1**) that is configured with the adaptation. For calls made to the VoIP card, Communication Manager will route the call over the appropriate SIP trunk (see **Section 5.8**). For outgoing calls from the VoIP card, the appropriate SIP entity will service the call as specified in the application sequence (see **Section 6.6.1**) configured in the VoIP card SIP user (see **Section 6.7**).

2.3. Support

For technical support and information on Symetrix 2 Line VoIP Interface Card, contact Symetrix customer support at:

- Phone: 1 (425) 778-7728
- Website: http://www.symetrix.co/support/
- Email: support@symetrix.co

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.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server (not shown in figure).
- 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 9600 and 96x1 Series H.323 and SIP Deskphones.
- Symetrix 2 Line VoIP Interface Card installed in Symetrix Radius AEC.

The Symetrix 2 Line VoIP Card registered with Session Manager as a SIP endpoint and was configured as Off-PBX Stations (OPS) on Communication Manager.

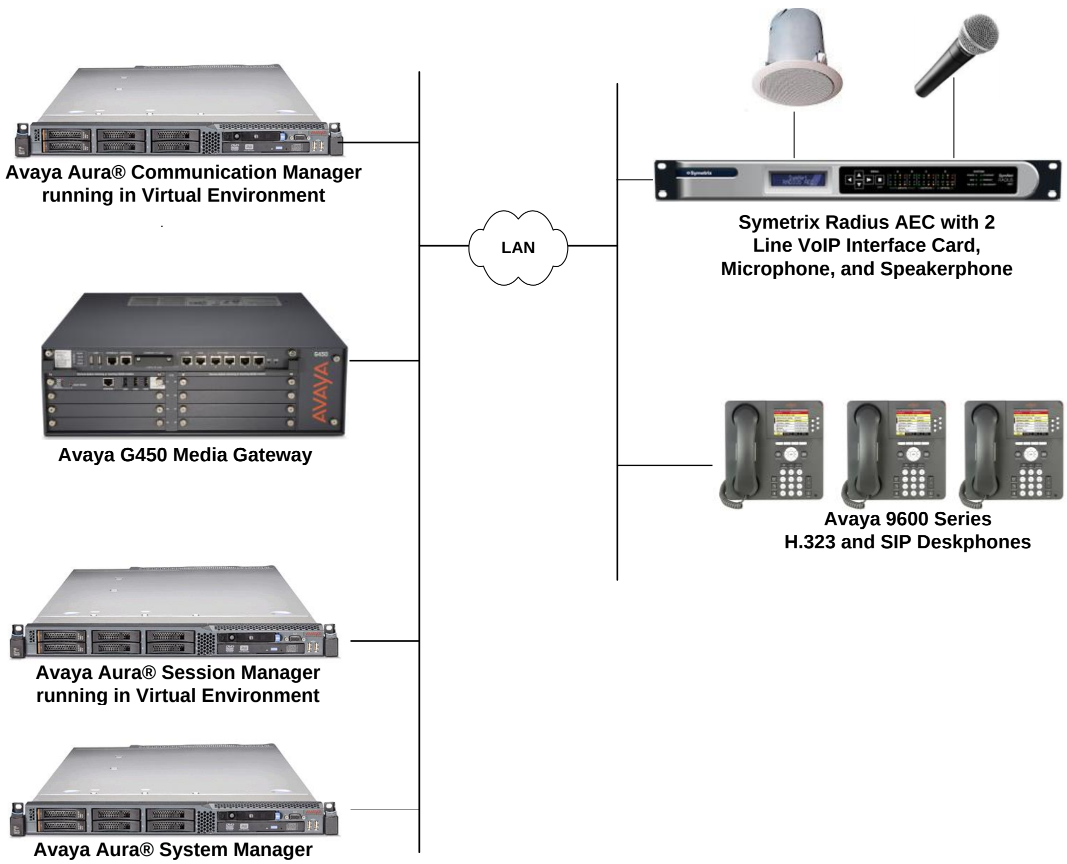


Figure 1: Avaya SIP-based Network and Symetrix Radius AEC with Symetrix 2 Line VoIP Interface Card

4. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	7.0.1.1 FP1 SP1 (R017x.00.0.441.0 with Patch 23169)
Avaya Aura® Media Server	7.7.0.226
Avaya Aura® Session Manager	7.0.1.1 (7.0.1.1.701114)
Avaya Aura® System Manager	7.0.1.1 (Build No. 7.0.0.016266 Software Update Revision No: 7.0.1.1.065378 Service Pack 1)
Avaya 9600 Series IP Phones	3.260A (H.323)
Avaya 96x1 Series IP Phones	7.0.1.1.5 (SIP)
Symetrix 2 Line VoIP Interface Card	6.16
Symetrix Radius AEC	5.4
Symetrix Composer	5.4

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager through the System Access Terminal (SAT). The procedures include the following areas:

- Verify License
- Administer IP Node Names
- Administer IP Codec Set
- Administer IP Network Region
- Administer SIP Trunks to Session Manager
- Configure Private Numbering
- Administer SIP Stations
- Administer AAR Call Routing

Important Note: For this compliance test, two SIP trunks between Communication Manager and Session Manager were established, one dedicated for calls to Symetrix 2 Line VoIP Card and another one for all other calls. The rationale for this is explained in **Section 2.2**. The difference between these two SIP trunks is that each uses a different SIP port and the SIP trunk used for calls to the 2 Line VoIP Interface Card has an adaptation rule applied to it on Session Manager.

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: V17                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
Platform: 28                                Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 6400  69
                                      Maximum Stations: 2400  21
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  0
                    Maximum Off-PBX Telephones -   OPS: 9600  13
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

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

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  0
           Maximum Concurrently Registered IP Stations: 2400  4
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  2
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  20
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    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-asm*). 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-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 5  of 5    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 Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the 2 Line VoIP Interface Card. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. For the compliance test, G.711MU, G.729, and G.722 codecs were used.

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

                          IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2:
```

5.4. Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, IP-IP Direct Audio (shuffling) is enabled to allow audio traffic to be sent directly between the 2 Line VoIP Interface Card and IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Media Server. The IP Network Region form also specifies the IP Codec Set to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

```
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: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      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
```

5.5. Administer SIP Trunks to Session Manager

As mentioned earlier, two SIP trunks between Communication Manager and Session Manager are required, one dedicated for calls to the Symetrix 2 Line VoIP Card and another one for all other calls. The difference between these two SIP trunks is that each will use a different SIP port and the SIP trunk used for calls to the 2 Line VoIP Interface Card will have an adaptation rule applied to it on Session Manager. The rationale for this is explained in **Section 2.2**. Basically, two SIP trunks are created so that an adaptation rule can be applied for calls to the Symetrix 2 Line VoIP Card only without affecting all other SIP calls on the Avaya SIP network.

5.5.1. SIP Trunk for Calls to Symetrix 2 Line VoIP Card

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 *tcp*.
- Specify the Ethernet processor (*procr*) of Communication Manager and 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 in **Section 5.2**.
- Ensure that the TCP port value of *5062* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields. Calls established with the 2 Line VoIP Interface Card should use a different SIP port than all other calls.
- 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*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- 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 20                                      Page   1 of   2
                                SIGNALING GROUP

 Group Number: 20              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM
 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: 5062                Far-end Listen Port: 5062
                                        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? n
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 the 2 Line VoIP Card. 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. Accept the default values for the remaining fields.

```
add trunk-group 20                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 20                   Group Type: sip          CDR Reports: y
  Group Name: Symetrix                    COR: 1        TN: 1        TAC: 1020
   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: 20
                                                 Number of Members: 10
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

```
change trunk-group 20                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

5.5.2. SIP Trunk for All Other Calls to Session Manager

Similar to the previous section, another SIP signaling group and SIP trunk group are required for routing all other SIP calls to Session Manager. This SIP trunk is not used to route calls to the 2 Line VoIP Interface Card. The configuration of the SIP signaling group is exactly the same as in the previous section, except that it must use a different SIP port. For this compliance test, signaling group 10 was created, which was configured to use SIP port 5060. In addition, trunk group 10 was also created similar to the one in the previous section, except that it was configured to use signaling group 10.

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' whose calls are routed over any trunk group, including SIP trunk groups 10 and 20, have the extension sent to Session Manager.

```
change private-numbering 0                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  7                                            5     Total Administered: 1
                                                          Maximum Entries: 540
```

The **Numbering – Public/Unknown Format** form was also configured as shown below.

```
change public-unknown-numbering 0                                   Page   1 of   2
                      NUMBERING - PUBLIC/UNKNOWN FORMAT
                                           Total
Ext Ext             Trk       CPN             CPN
Len Code            Grp(s)    Prefix          Len
                                                    Total Administered: 1
 5  7                                         5        Maximum Entries: 240

                                                    Note: If an entry applies to
                                                    a SIP connection to Avaya
                                                    Aura(R) Session Manager,
                                                    the resulting number must
                                                    be a complete E.164 number.

                                                    Communication Manager
                                                    automatically inserts
                                                    a '+' digit in this case.
```

5.7. Administer SIP Stations

A SIP station is configured for the 2 Line VoIP Interface Card. The **Type** field should be set to *9600SIP* and the system will assign an appropriate port after the station has been added. During initial creation of the station, the **Port** field is set to *IP*. A descriptive **Name** is also configured. The SIP station was configured automatically by System Manager as described in **Section 6.7** and it is shown below as it would appear on Communication Manager.

```
display station 78020                                           Page   1 of   6
                                     STATION

Extension: 78020                        Lock Messages? n               BCC: 0
     Type: 9600SIP                      Security Code:                  TN: 1
     Port: S00028                     Coverage Path 1:                 COR: 1
     Name: 78020, Symetrix            Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 78020
Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english               Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n

                                              Customizable Labels? y
```

Configure the **Stations with Off-PBX Telephone Integration** form so that calls destined for the 2 Line VoIP Interface Card are routed over trunk group 20 (configured in **Section 5.5.1**) to Session Manager, which will then route the call to the 2 Line VoIP Interface Card. On this form, specify the extension of the SIP endpoint and set the **Application** field to *OPS*. The **Phone Number** field is set to the digits to be sent over the SIP trunk. In this case, the SIP extension configured on Session Manager also matches the extension of the corresponding station on Communication Manager. However, this is not a requirement. Finally, the **Trunk Selection** field is set to *aar*. This field specifies Auto Alternate Routing (AAR) routing. In this case, the **Trunk Selection** field would be set to *aar* to trigger AAR routing. Configuration of the **AAR Analysis** and **Route Pattern** forms would also be required (see **Section 5.8**). This form was also configured through System Manager.

```
change off-pbx-telephone station-mapping 78020                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 78020            OPS           -     78020            aar        1
```

5.8. 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 route calls to the 2 Line VoIP Interface Card (i.e., extension *78020*) to route pattern 20 as shown below. All other calls, where digits beginning with "78" are dialed, are routed over route pattern 10 also shown below.

```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 2

          Dialed            Total      Route     Call    Node   ANI
          String          Min  Max   Pattern    Type    Num    Reqd
    78                    5    5      10        lev0           n
    78020                 5    5      20        lev0           n
```

Configure a preference in **Route Pattern** 20 to route calls over SIP trunk group 20 as shown below. This routes calls to the 2 Line VoIP Card.

```
change route-pattern 20                                         Page   1 of   3
                    Pattern Number: 20     Pattern Name: Symetrix
    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: 20   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
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below. This route pattern is used for all calls, except calls to the 2 Line VoIP Card.

```
change route-pattern 10                                         Page   1 of   3
                    Pattern Number: 10     Pattern Name: 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
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- Adaptation
- Communication Manager SIP Entities
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Define Communication Manager as Administrable Entity (i.e., Managed Element)
- Application Sequences
- SIP User for 2 Line VoIP Interface Card

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials.

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of the adaptation, SIP entities, entity links, and a SIP user for the 2 Line VoIP Interface Card.

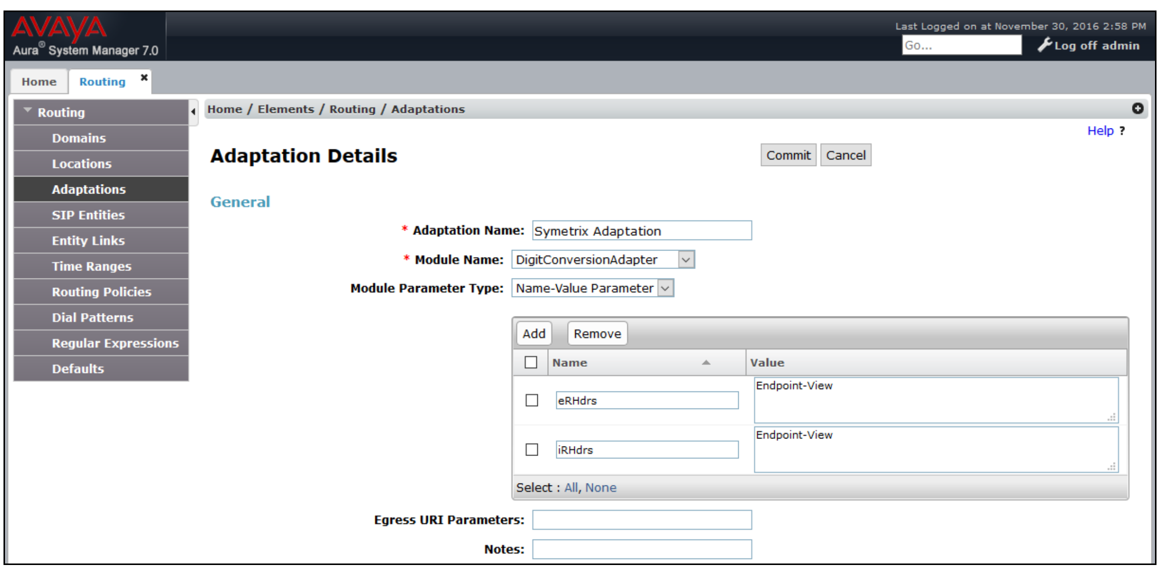
For this compliance test, two SIP trunks and two entity links for Communication Manager were created so that an adaptation can be applied to one SIP entity but not the other. The adaptation will remove the Endpoint-View header in the SIP INVITE for calls involving the 2 Line VoIP Interface Card. This adaptation should not be applied for all other calls; hence, the second SIP entity and entity link. See the observation in **Section 2.2** for more details.

6.1. Add Adaptation

Session Manager can be configured with Adaptations that can modify SIP messages before or after routing decisions have been made; for example, removing the Endpoint-View header in a SIP INVITE message. To create an **Adaptation** that will be applied to the Communication Manager SIP entity in **Section 6.2.1**, navigate to **Elements → Routing → Adaptations** and click on the **New** button (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields.

- **Adaptation Name:** Enter a descriptive name for the Adaptation (e.g., *Symetrix Adaptation*).
- **Module Name:** Select **DigitConversionAdapter**.
- **Module Parameter Type:** Select **Name-Value Parameter**. The section will expand with an area to enter **Name** and **Value** pairs. Click **Add**. To remove headers on the egress side of Session Manager (i.e., towards Communication Manager) enter the keyword **eRHdrs** in the **Name** field and *Endpoint-View* in the **Value** field to remove this header. Click **Add** again. To remove headers on the ingress side of Session Manager (i.e., from Communication Manager) enter the keyword **iRHdrs** in the **Name** field and *Endpoint-View* in the **Value** field to remove this header.



6.2. Add SIP Entities

In the sample configuration, two SIP Entities are added for Communication Manager, one will be used be used for calls involving the 2 Line VoIP Interface Card and another one will be used for all other calls.

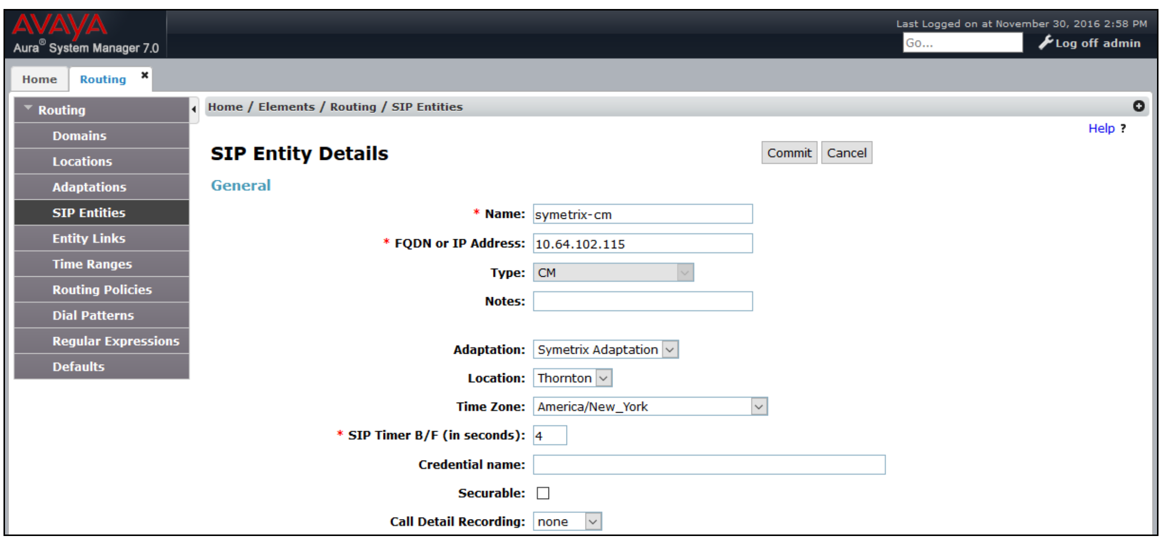
6.2.1. SIP Entity for Avaya Aura® Communication Manager for Calls with Symetrix 2 Line VoIP Interface Card

A SIP Entity must be added for Communication Manager for calls with the 2 Line VoIP Interface Card. This SIP entity will have an adaptation rule to remove the Endpoint-View header in SIP INVITE messages. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface of Communication Manager.
- **Type:** Select *CM*.
- **Adaptation :** Select the Adaptation configured in **Section 6.1**.
- **Location:** Select the appropriate location.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save the SIP Entity definition.



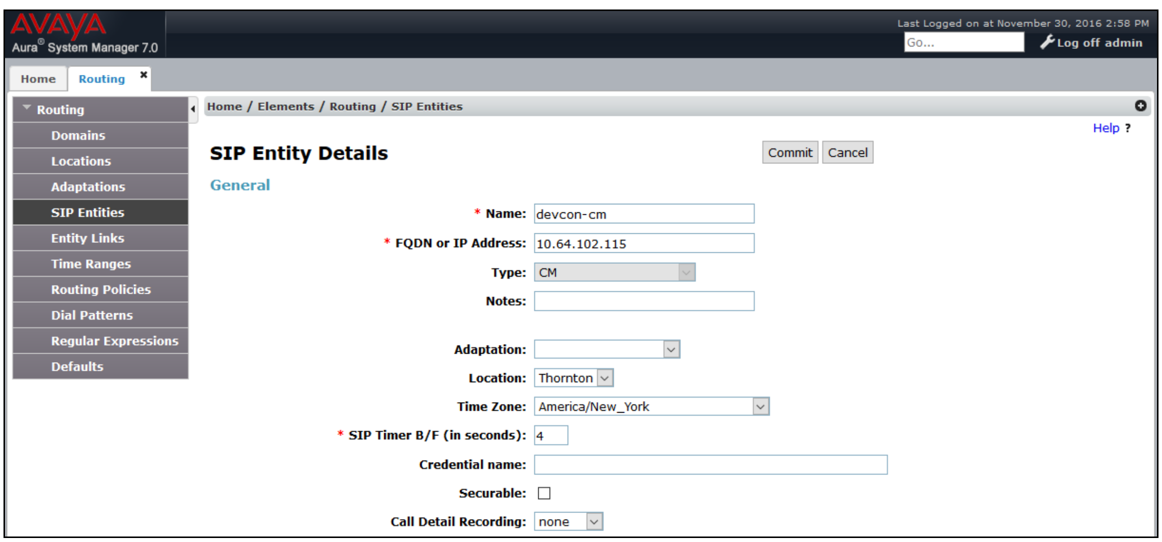
6.2.2. SIP Entity for Avaya Aura® Communication Manager for All Other Calls

Another SIP Entity for Communication Manager is created. This one is used for all other calls, except for calls with the 2 Line VoIP Interface Card. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface of Communication Manager.
- **Type:** Select *CM*.
- **Adaptation:** No adaptation is used (i.e., leave blank).
- **Location:** Select the appropriate location.
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click Commit to save the SIP Entity definition.



6.3. Add Entity Links

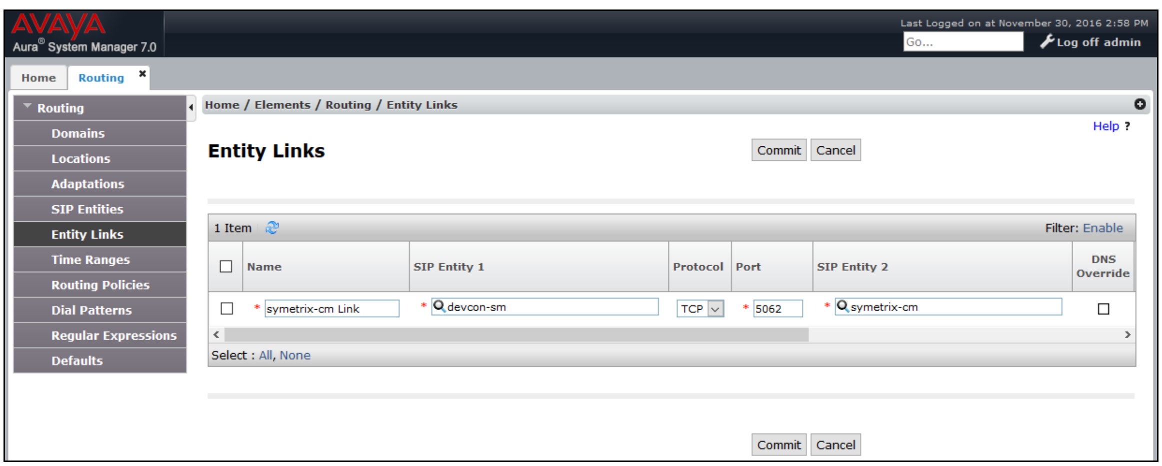
This section covers the configuration of Entity Links for Communication Manager. Two entity links are configured. One entity link will be used for calls with the 2 Line VoIP Interface Card and will be configured to use SIP port 5062. This entity link will specify that SIP entity configured in **Section 6.2.1**, which applies the adaptation to remove the Endpoint-View header in SIP INVITE messages. The other entity link will be used for all other calls and will be configured to use standard SIP port 5060. This entity link will specify the SIP entity configured in **Section 6.2.2**, which will not manipulate any SIP messages (i.e., no adaptation will be applied).

6.3.1. Communication Manager Entity Link for Calls with Symetrix 2 Line VoIP Interface Card

The SIP trunk from Session Manager to Communication Manager is described by an Entity Link. This entity link will be used for calls with the 2 Line VoiP Interface Card. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *symetrix-cm link*).
- **SIP Entity 1:** Select Session Manager.
- **Protocol:** Select the appropriate protocol (e.g., *TCP*).
- **Port:** Port number to which the other system sends SIP requests. Port 5062 is used for calls with the 2 Line VoIP Interface Card.
- **SIP Entity 2:** Select the SIP entity for Communication Manager configured in **Section 6.2.1**.
- **Port:** Port number on which the other system receives SIP requests. Port 5062 is used for calls involving the 2 Line VoIP Interface Card.
- **Connection Policy:** Select *Trusted*. **Note:** If Trusted is not selected, calls from the associated SIP Entity specified in **Section 6.2.1** will be denied.

Click **Commit** to save the Entity Link definition.

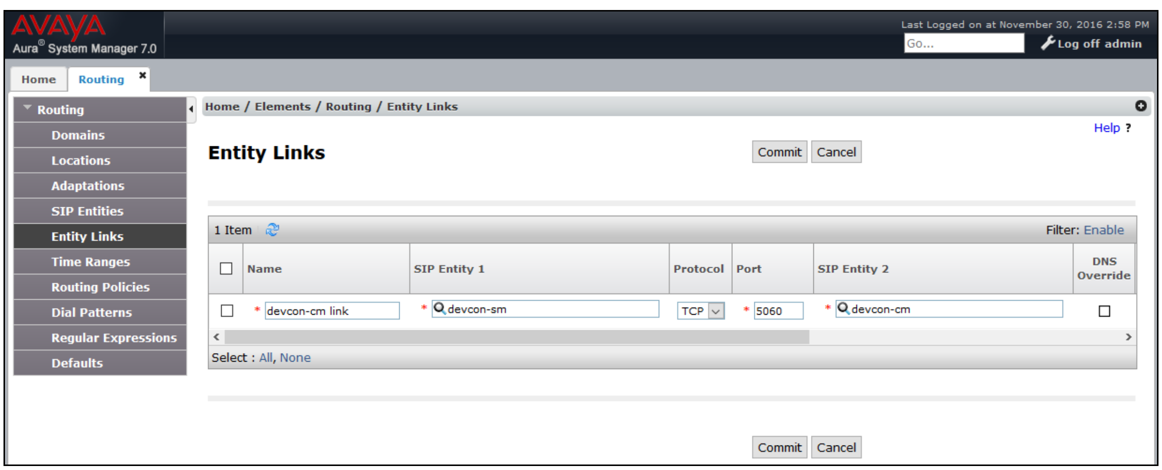


6.3.2. Communication Manager Entity Link for All Other Calls

The SIP trunk from Session Manager to Communication Manager is described by an Entity Link. This entity link will be used for all other calls, except for calls with the 2 Line VoiP Interface Card. To add an Entity Link, select Entity Links on the left and click on the New button (not shown) on the right. Fill in the following fields in the new row that is displayed:

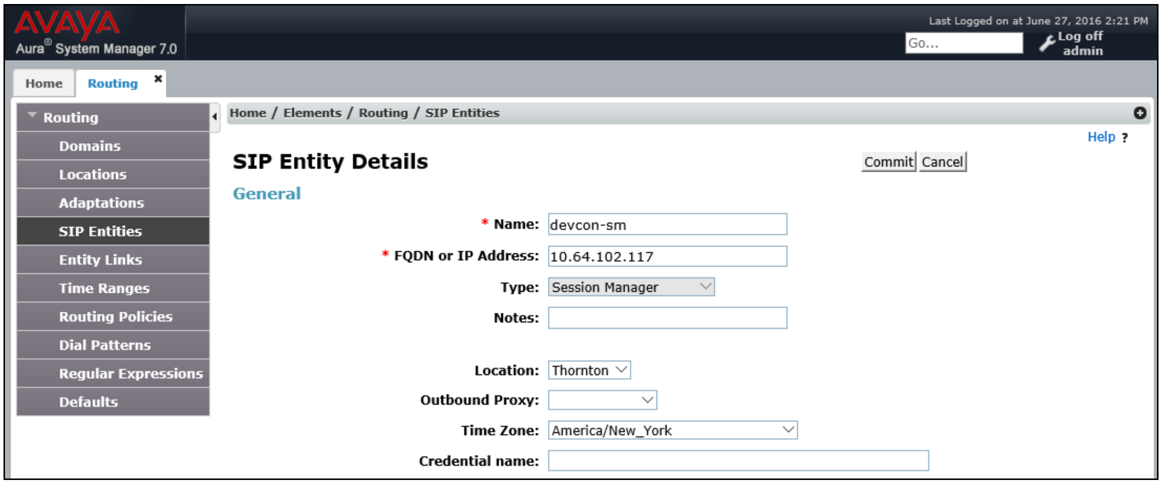
- **Name:** A descriptive name (e.g., *devcon-cm Link*).
- **SIP Entity 1:** Select Session Manager.
- **Protocol:** Select the appropriate protocol (e.g., TCP).
- **Port:** Port number to which the other system sends SIP requests. Port 5060 is used for all other calls, except calls with the 2 Line VoIP Interface Card.
- **SIP Entity 2:** Select the SIP entity for Communication Manager configured in **Section 6.2.2**.
- **Port:** Port number on which the other system receives SIP requests. Port 5060 is used for all other calls, except calls with the 2 Line VoIP Interface Card.
- **Connection Policy:** Selected *Trusted*. **Note:** If the link is not trusted, calls from the associated SIP Entity specified in **Section 6.2.2** will be denied.

Click **Commit** to save the Entity Link definition.

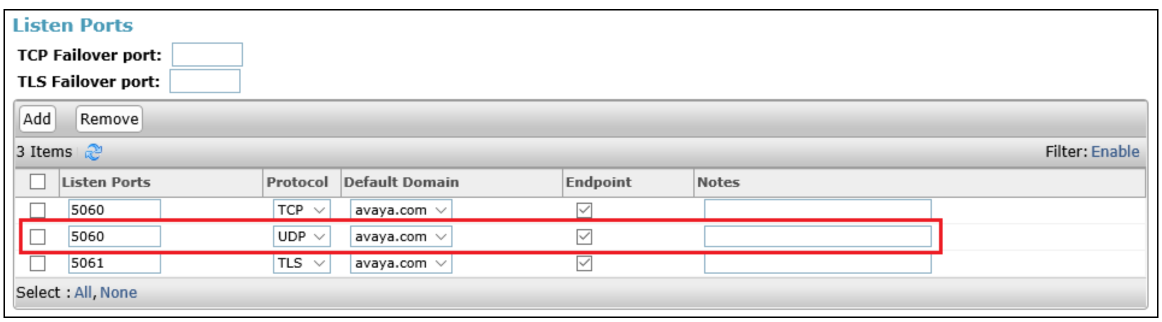


6.4. Set Network Transport Protocol for SIP Users

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



Scroll down to the **Listen Ports** section and verify that the transport network protocol used by the 2 Line VoIP Interface Card is specified in the list below. For the compliance test, UDP network transport was used, but TCP is also supported.



6.5. Define Communication Manager as Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

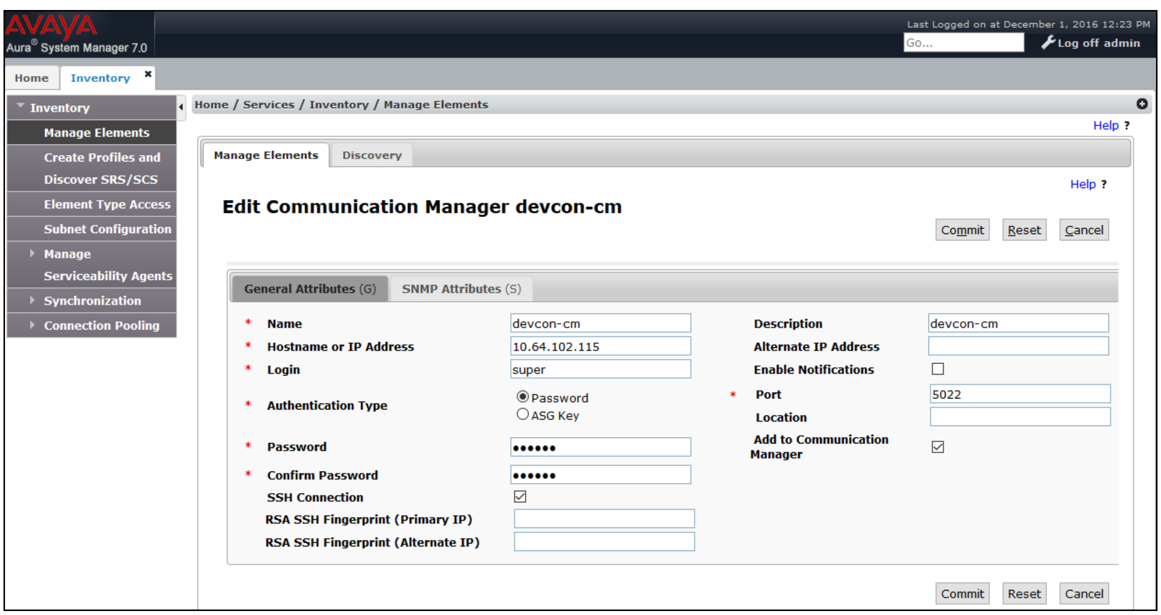
To define Communication Manager as a managed element, select **Services→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **Application Type** field that is displayed, select *CM*.

In the **New CM Instance** screen, first select Communication Manager as the Type (not shown), and then fill in the following fields as follows:

Under *General Attributes*:

- **Name:** Enter an identifier for Communication Manager.
- **Hostname or IP Address:** Enter the IP address of the administration interface for Communication Manager.
- **Login / Password:** Enter the login and password used for administration access.
- **Authentication Type:** Select Password.
- **SSH Connection:** Select checkbox.
- **Port:** Enter the port number for SSH administration access (5022).

Defaults can be used for the remaining fields. Click **Commit** to save the settings.



6.6. Add Application Sequences

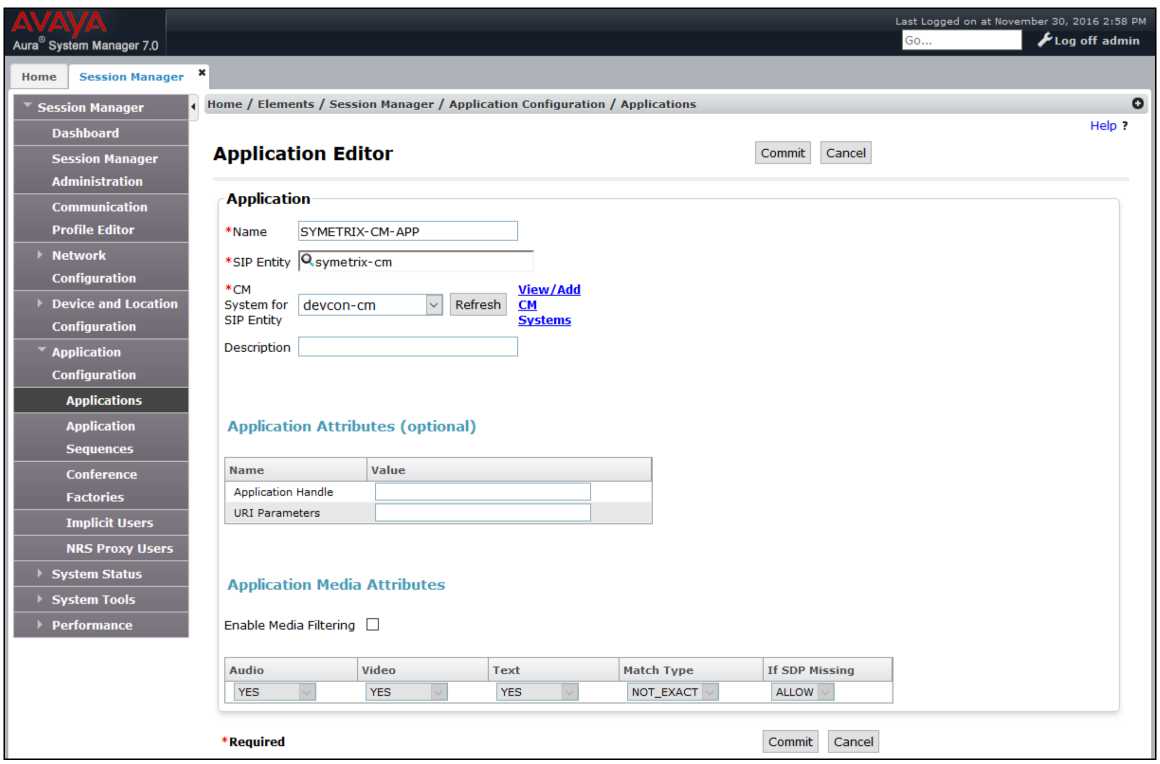
Two **Application Sequences** for Communication Manager are required for this solution, one for the 2 Line Voice Interface Card and another one for all other SIP endpoints.

6.6.1. Application Sequence for Symetrix 2 Line VoIP Interface Card

To define an application for Communication Manager (to be used for the 2 Line VoIP Interface Card), navigate to **Elements → Session Manager → Application Configuration → Applications** on the left and select **New** button (not shown) on the right. Fill in the following fields:

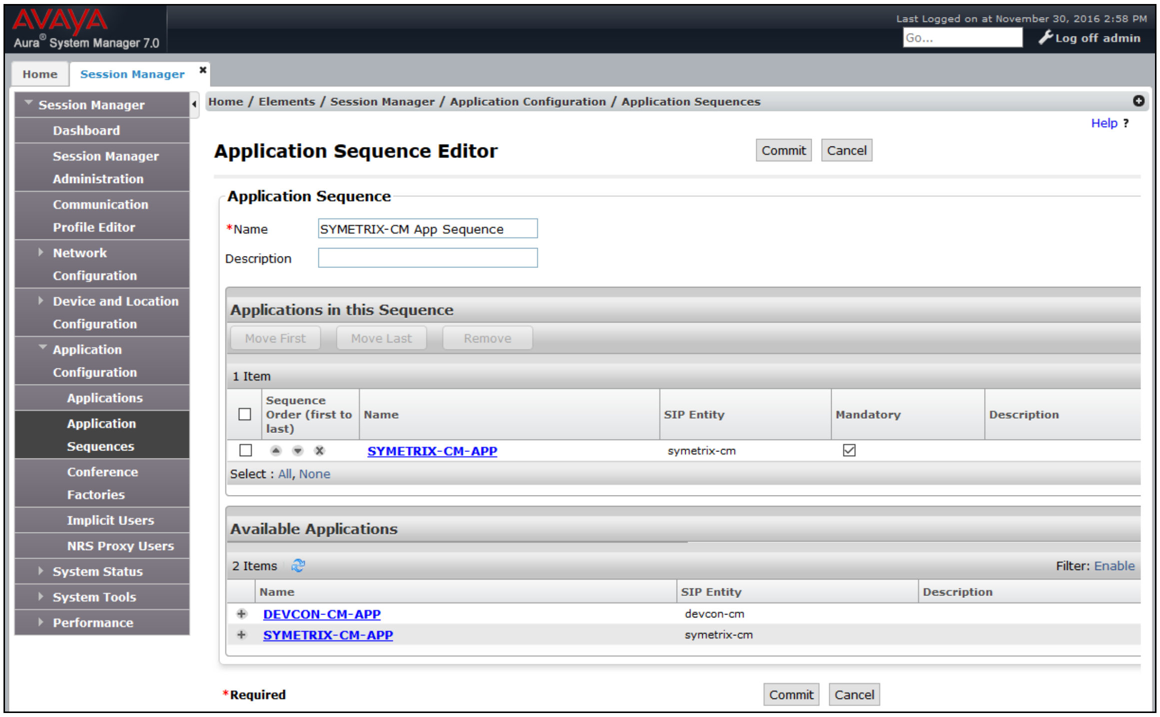
- **Name:** Enter name for application.
- **SIP Entity:** Select the Communication Manager SIP entity configured in **Section 6.2.1** (i.e., *symetrix-cm*)
- **CM System for SIP Entity:** Select the Communication Manager managed element configured in **Section 6.4**.

Click **Commit** to save the Application definition.



Next, define the Application Sequence for Communication Manager as shown below.

Verify a new entry is added (i.e., *SYMETRIX-CM-APP* configured above) to the **Applications in this Sequence** table and the **Mandatory** column is as shown below.



6.6.2. Application Sequence for All Other SIP Endpoints

The configuration of the **Application Sequence** for all other SIP endpoints is similar to **Section 6.6.1**, except that the **Application** will specify a different name (e.g., *DEVCON-CM-APP*) and use the Communication Manager SIP entity configured in **Section 6.2.2** and the **Application Sequence** will specify a different name (e.g., *DEVCON-CM App Sequence*) and the aforementioned **Application** (e.g., *DEVCON-CM-APP*) will be selected.

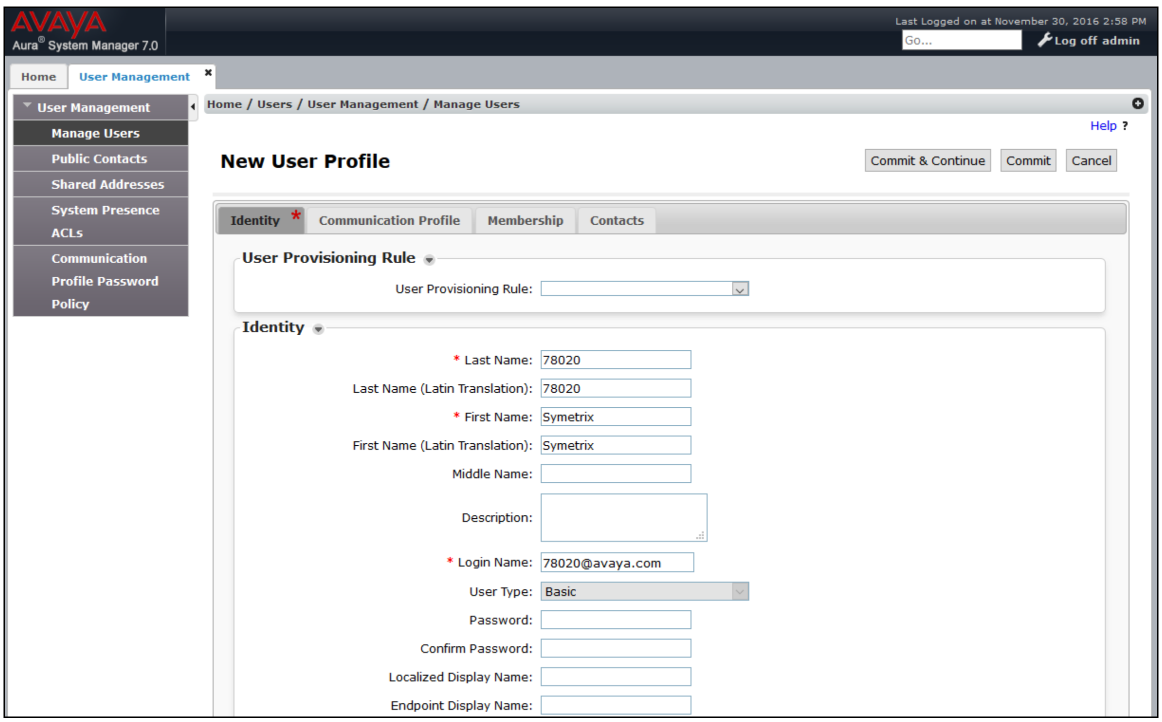
6.7. Configure SIP User

Add a SIP user for the 2 Line VoIP Interface Card. To add new SIP users, expand **Users** and select **Manage Users** from left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** tab of the new user form.

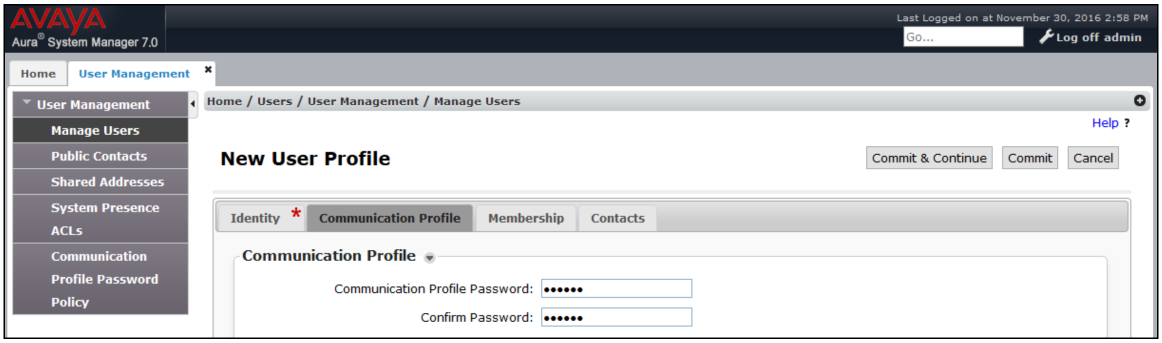
- **Last Name:** Enter the last name of the user (e.g., *78020*).
- **First Name:** Enter the first name of the user (e.g., *Symetrix*).
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., *78020@avaya.com*).

The screen below shows the information when adding a new SIP user to the sample configuration.



Select the **Communication Profile** tab and configure the following fields:

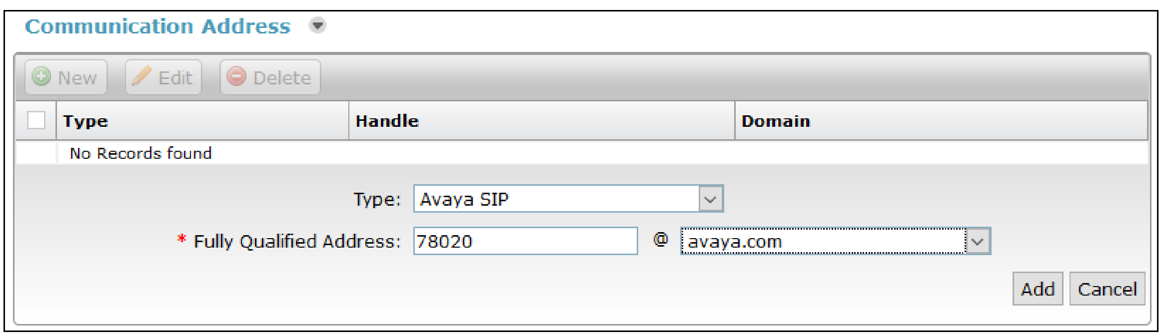
- **Communication Profile Password:** Enter the password which will used by the 2 Line VoIP Interface Card to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.



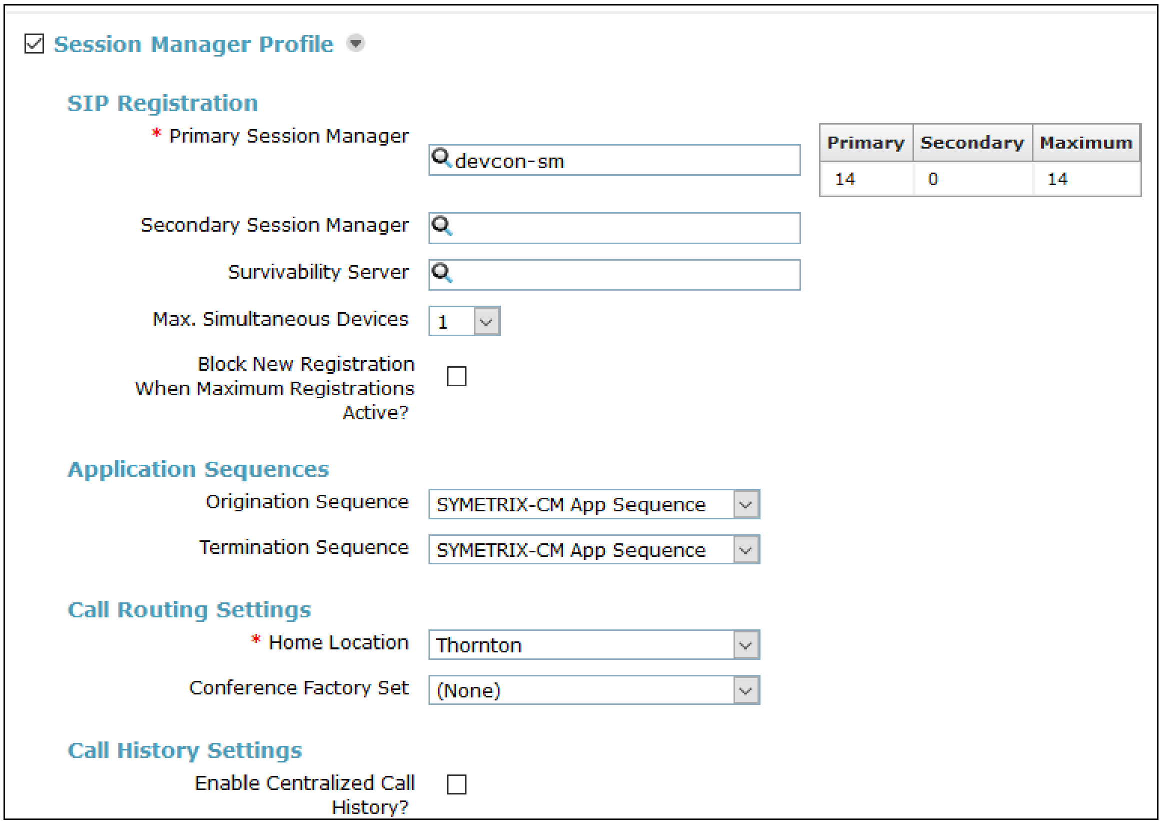
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.



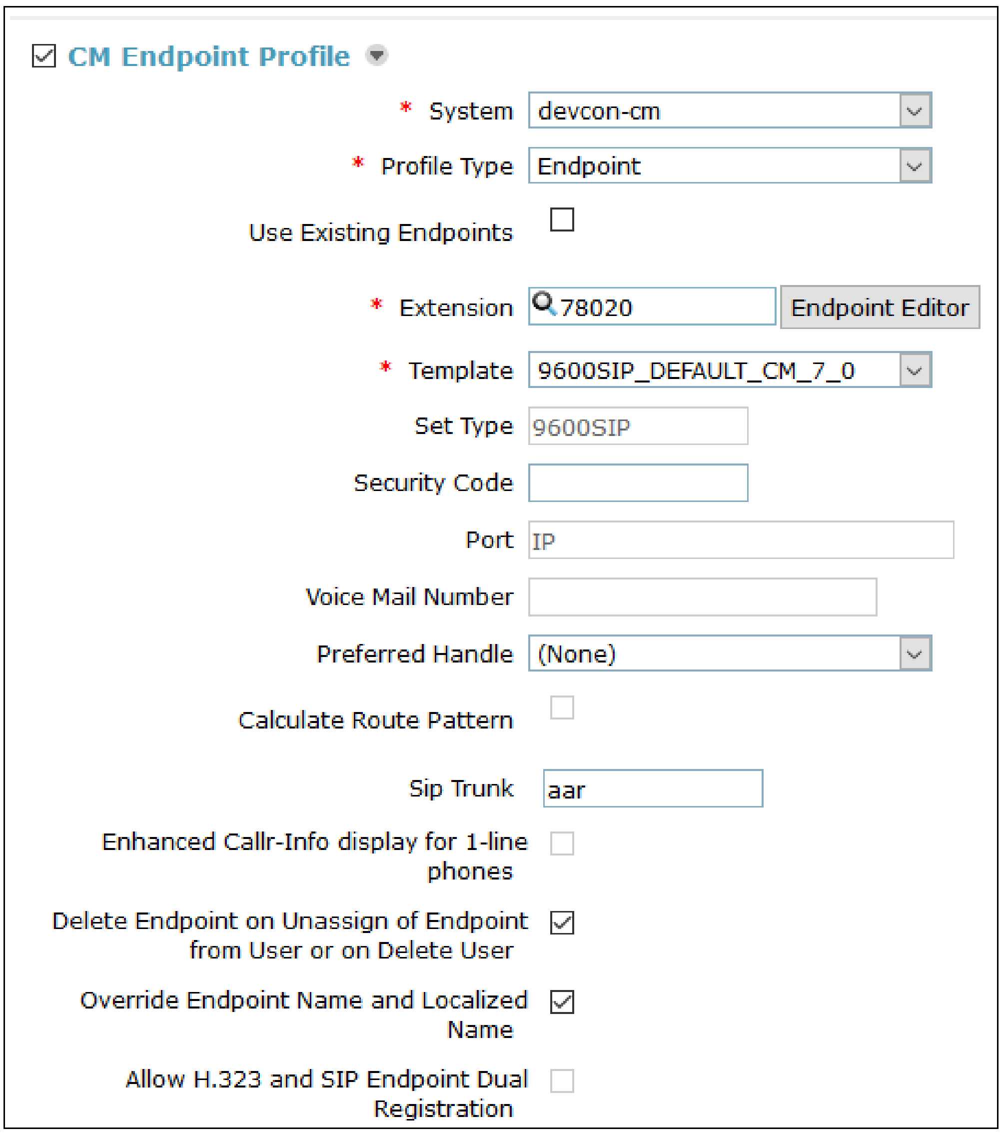
In the *Session Manager Profile* section, specify the Session Manager SIP entity for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 6.6.1** to both the originating and terminating sequence fields. This application sequence specifies the SIP entity configured in **Section 6.2.1** so that the Communication Manager SIP entity with the adaptation is used when placing outgoing calls from the 2 Line VoIP Interface Card. Set the **Home Location** field to the appropriate **Location**.



In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for 9600SIP.
- **Port:** Enter *IP*.
- **Sip Trunk:** Specify *AAR*.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Commit** (not shown) to add the SIP user.



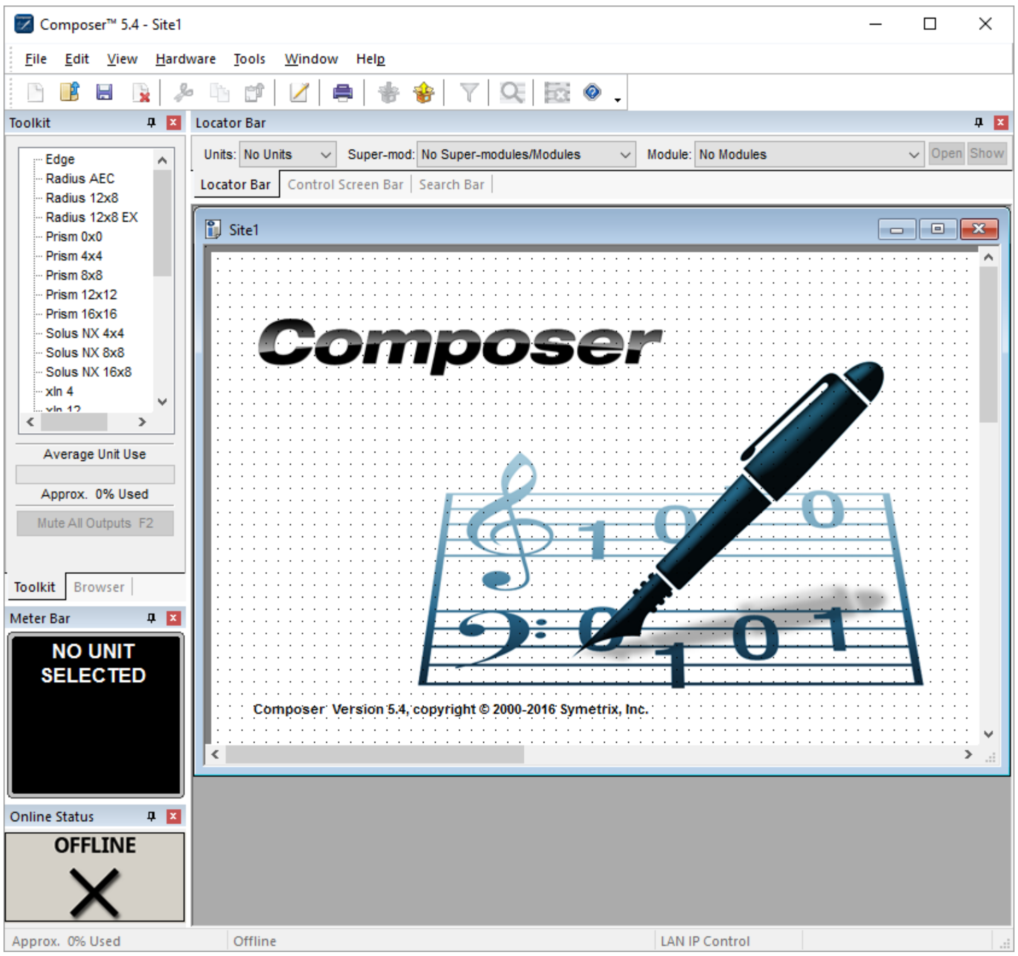
7. Configure Symetrix 2 Line VoIP Interface Card

This section covers the configuration of the 2 Line VoIP Interface Card using the Symetrix Composer application. The configuration covers the following areas:

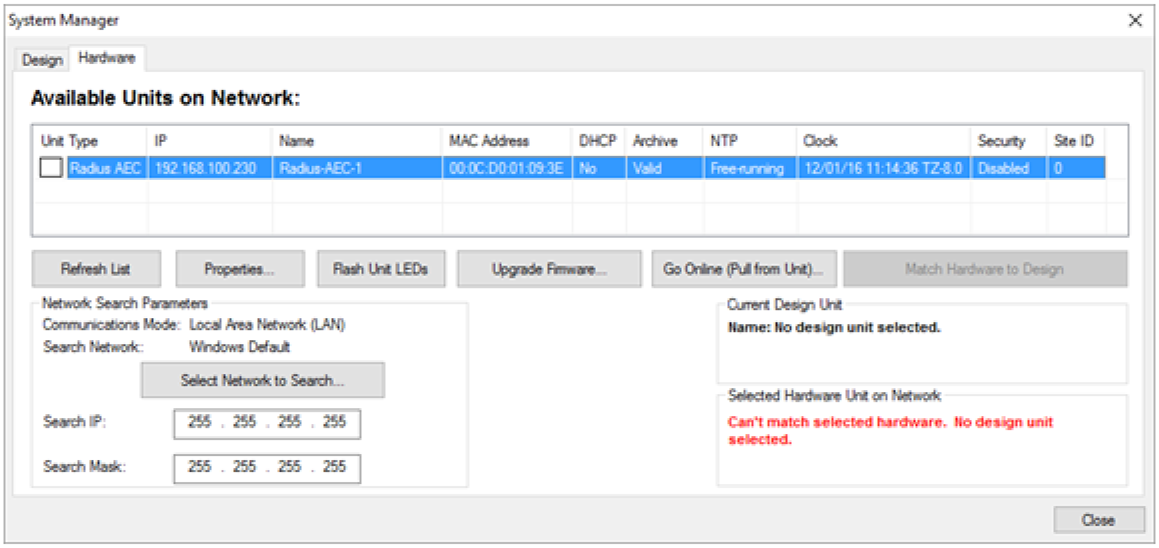
- Launch the Composer Application and Connect to Radius AEC
- Modify the IP Network Parameters of Radius AEC to Correspond to the Customer's Network
- Configure the 2 Line VoIP Interface Card, including:
  - IP Network Parameters
  - SIP Line
  - SIP/VoIP Parameters

7.1. Launch Composer and Connect to Symetric Radius AEC

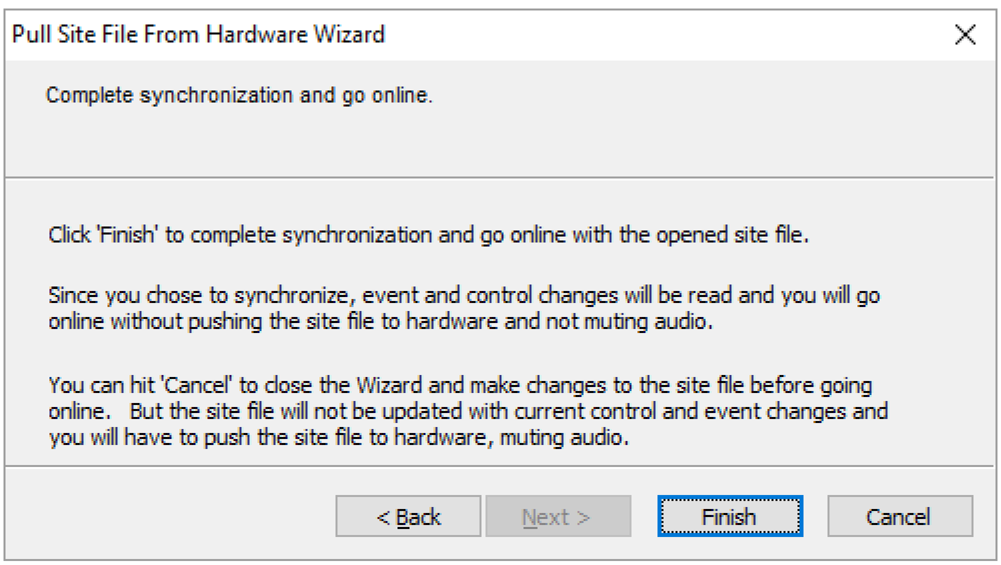
Start the **Composer** application to display the main window below.



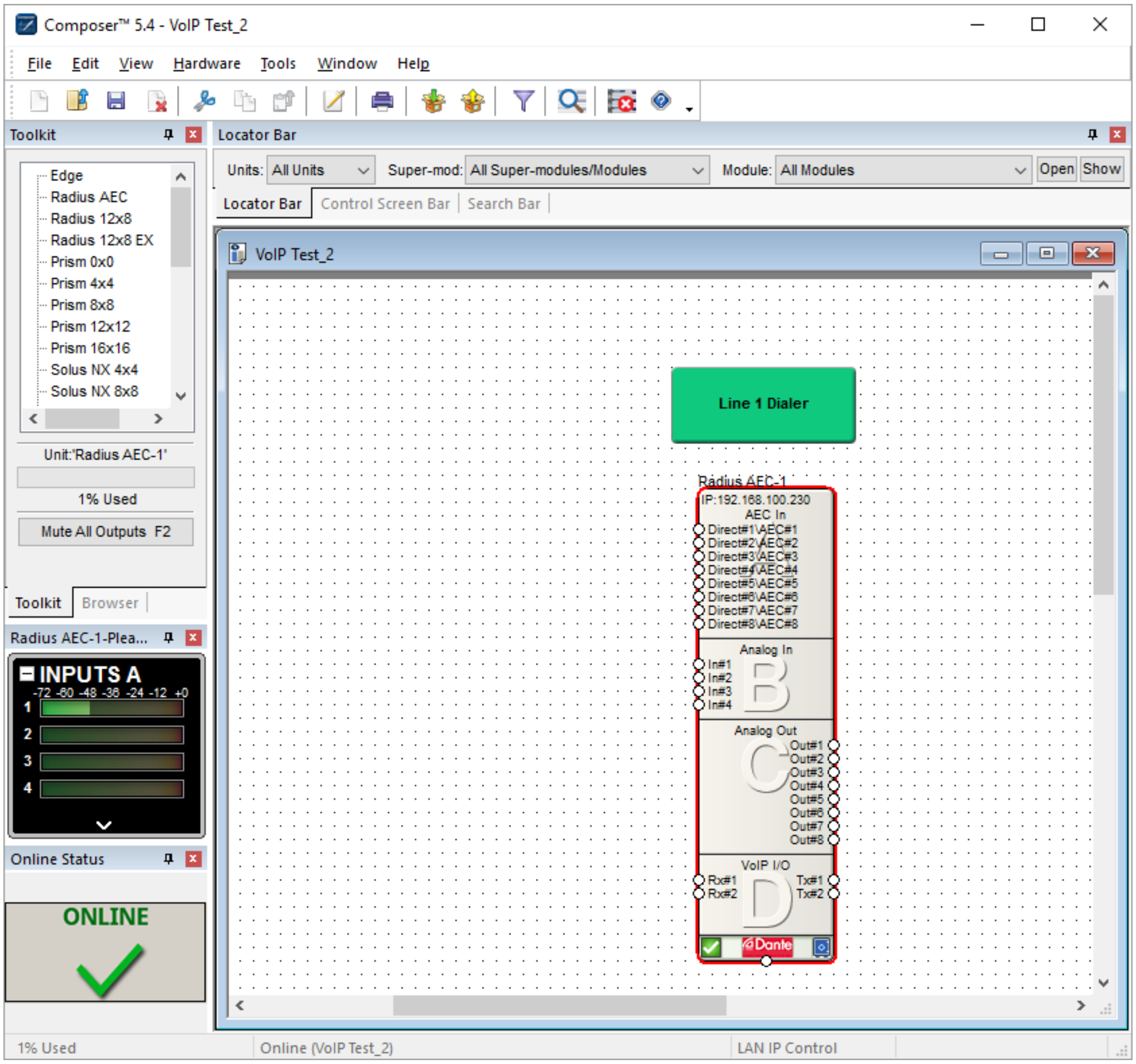
From **Composer**, select **Hardware → System Manager** to display the **Available Units on Network**. Select the Radius AEC and click on **Go Online (Pull from Unit)** to connect to the system as shown below.



In the subsequent prompts, click on **Next** or **OK** until the following window is displayed. Click **Finish**.

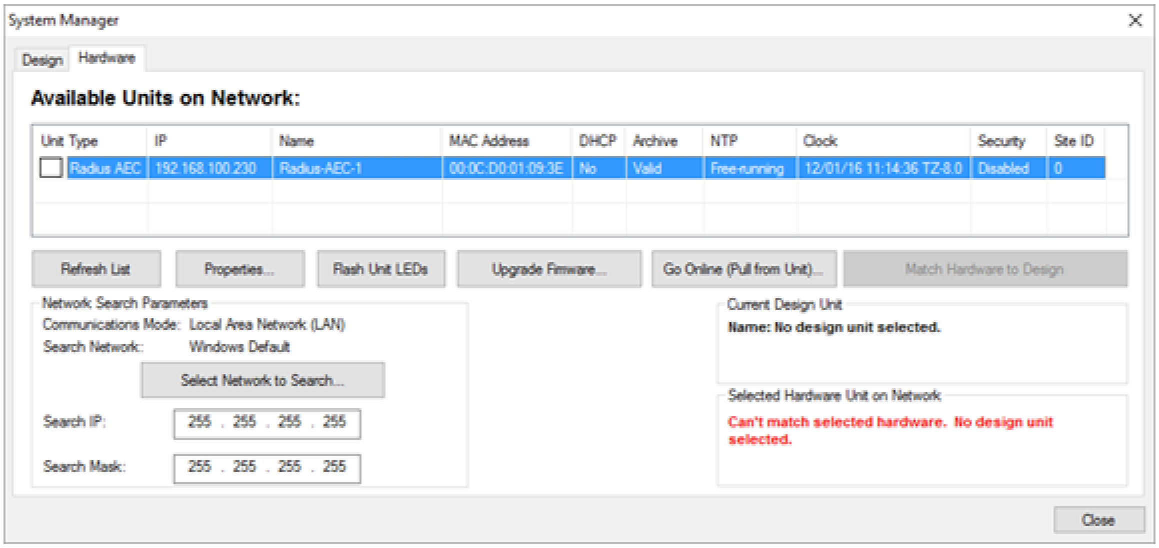


Composer will update the connection status with a green checkmark and display a graphical representation of the Radius AEC unit as shown below. The user is now connected to Radius AEC.

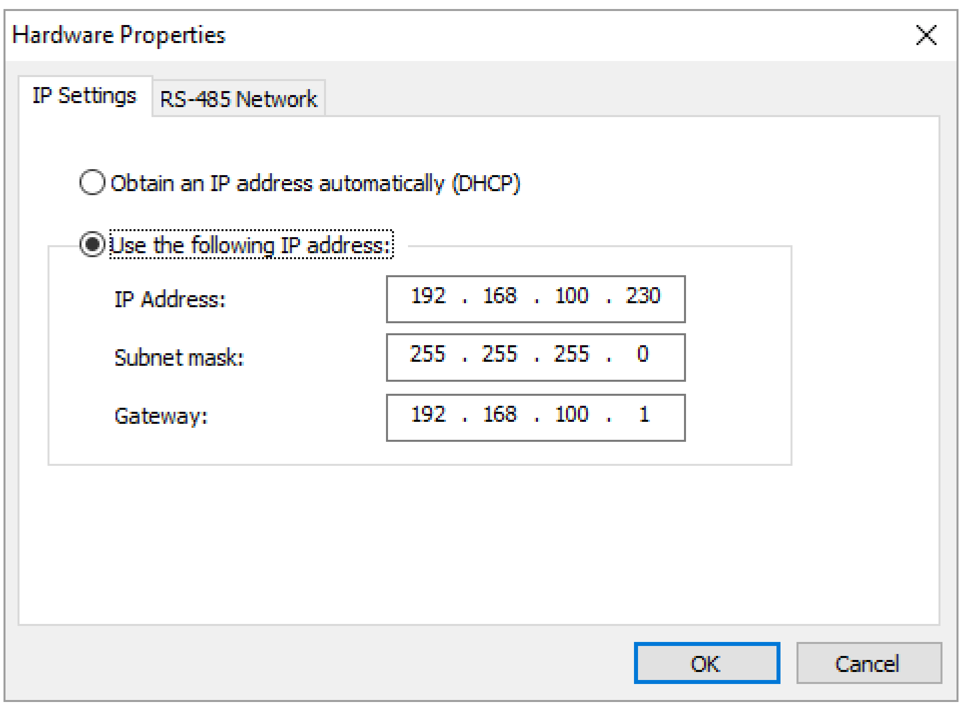


7.2. Modify the IP Network Parameters of Radius AEC

Modify the IP address of Radius AEC, if necessary, to correspond to the customer's network. From the **Composer** window displaying the **Available Units on Network** shown below, click on the **Properties** button.

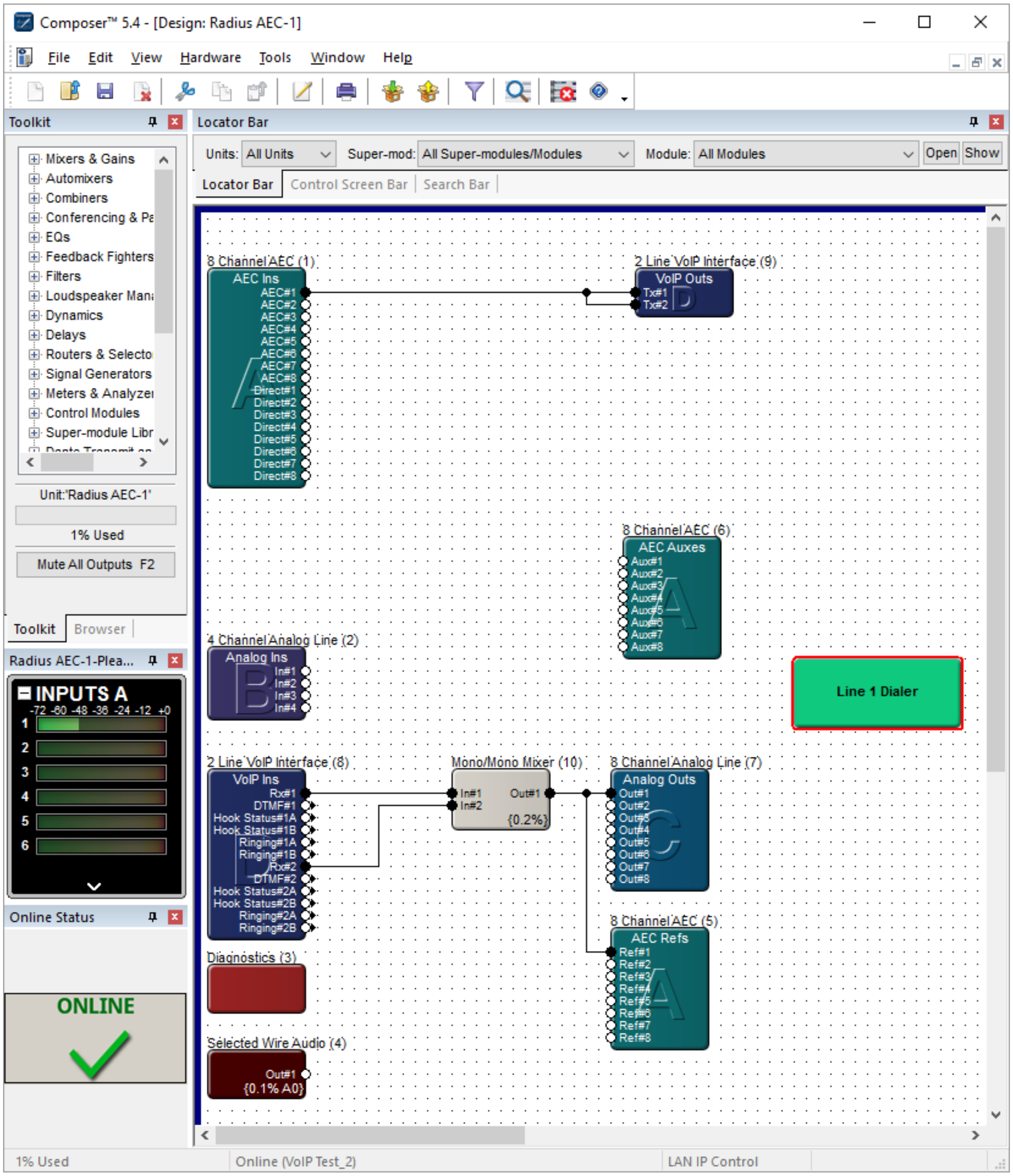


In the **Hardware Properties** window, modify the IP network parameters as necessary. Click **OK**.

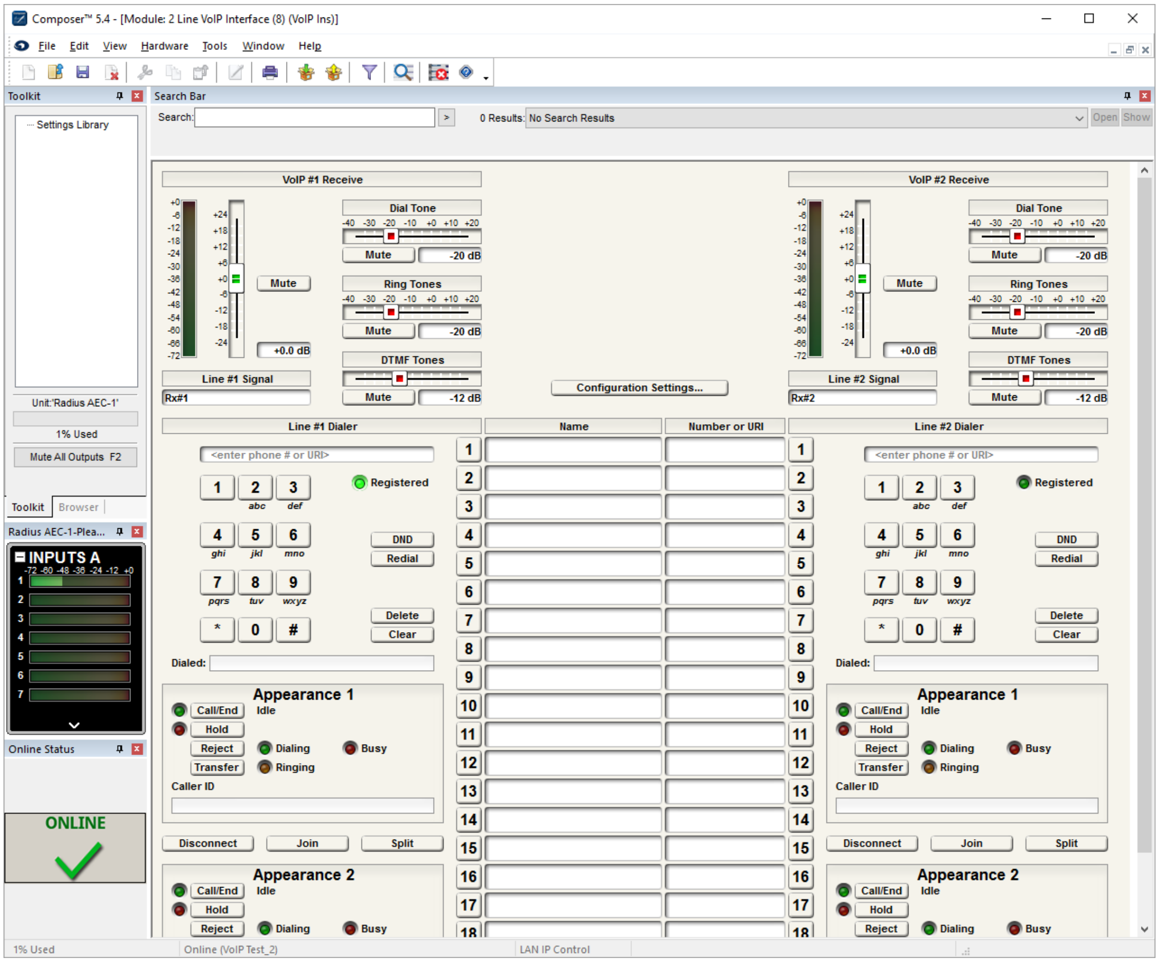


7.3. Configure Symetrix 2 Line VoIP Interface Card

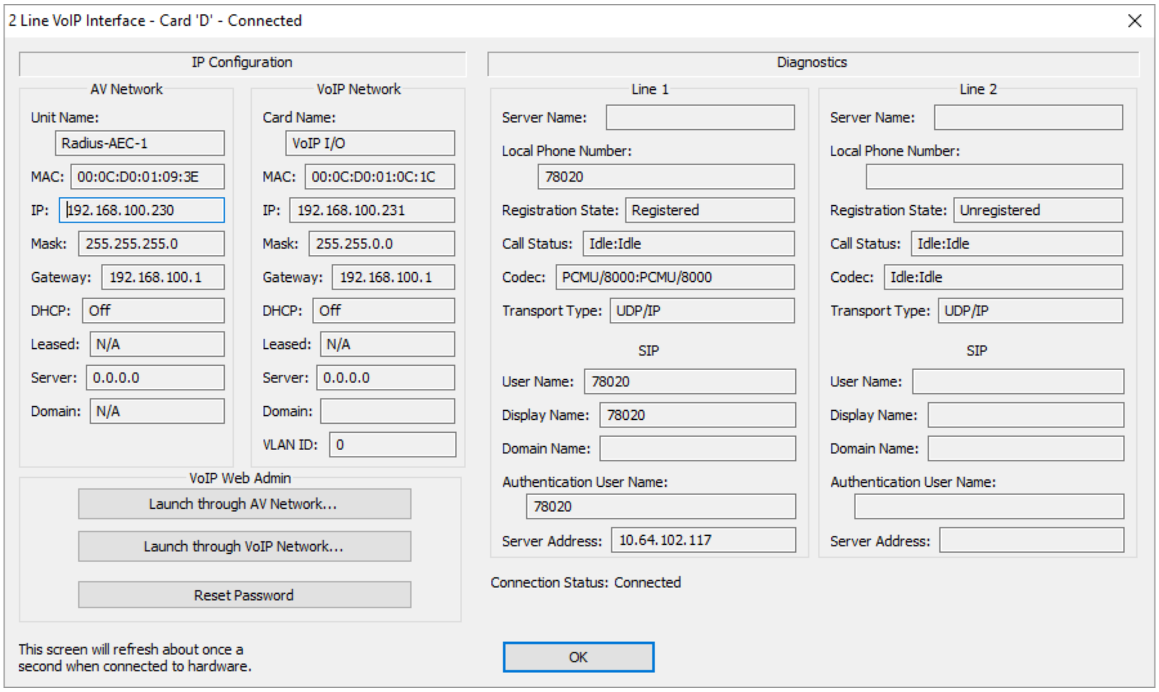
From **Composer**, double-click on the Radius AEC graphic to display the Radius AEC hardware configuration shown below.



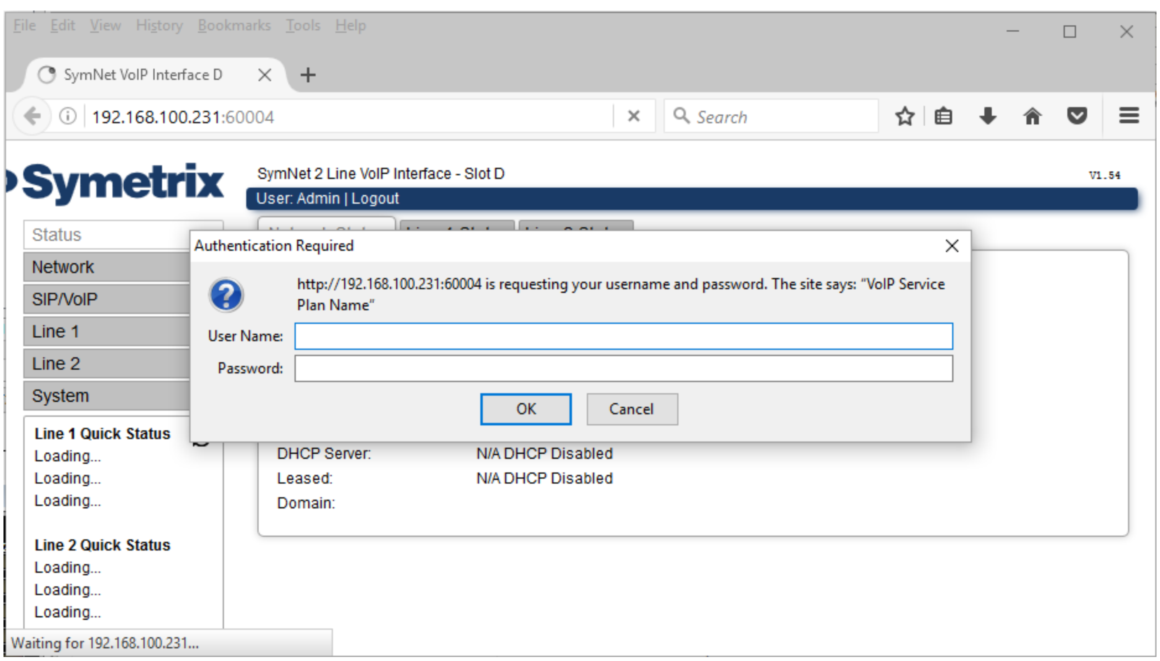
Next, double-click on the **2 Line VoIP Interface** graphic in the window above to display the following window. Click on the **Configuration Settings** button in the window below.



From the following window, click on the **Launch through VoIP Network** to open the **VoIP Web Admin** in an internet browser.



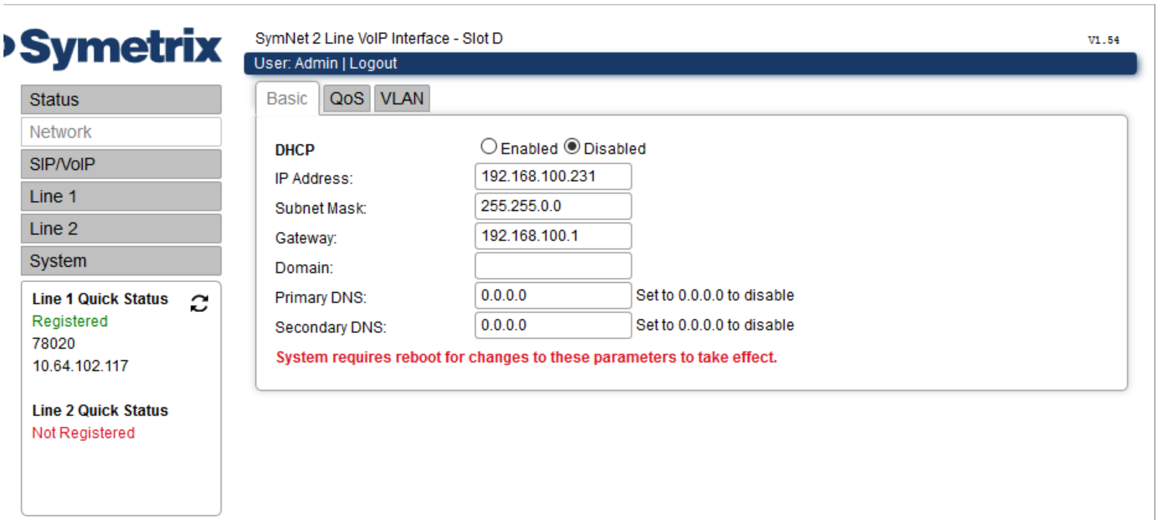
Log into the **VoIP Web Admin** interface.



Once logged in, the 2 Line VoIP Interface Card may be configured via the VoIP Web Admin interface, including the VoIP card IP address, SIP Line, and other SIP/VoIP Parameters.

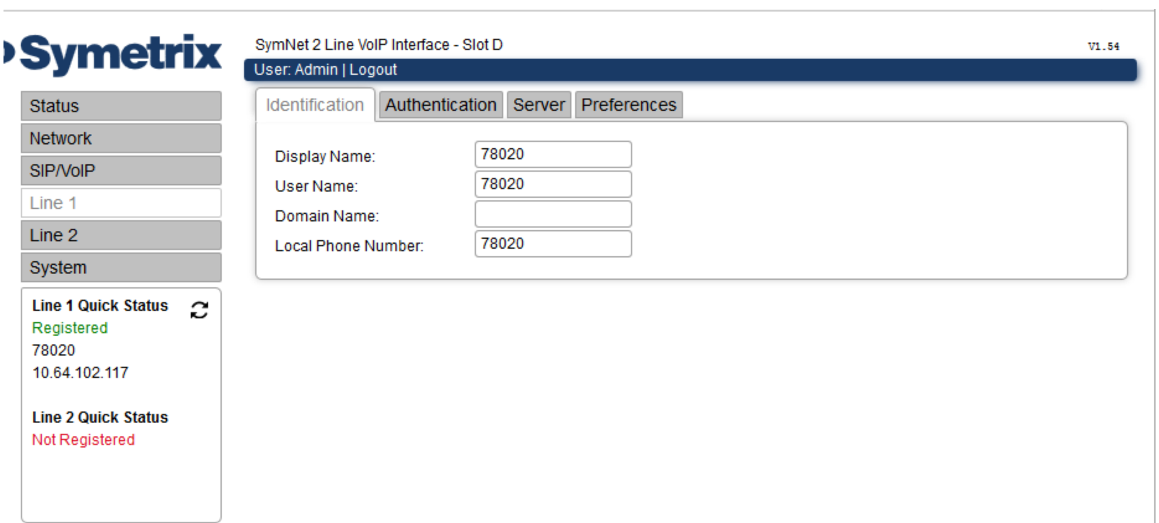
7.3.1. Modify IP Address of Symetrix 2 Line VoIP Interface Card

Modify the IP address of the 2 Line VoIP Interface Card, if necessary, to correspond to the customer's network. Select the **Network** option from the left pane and configure the IP network parameters to the desired values as shown below. Click **Save** (not shown).

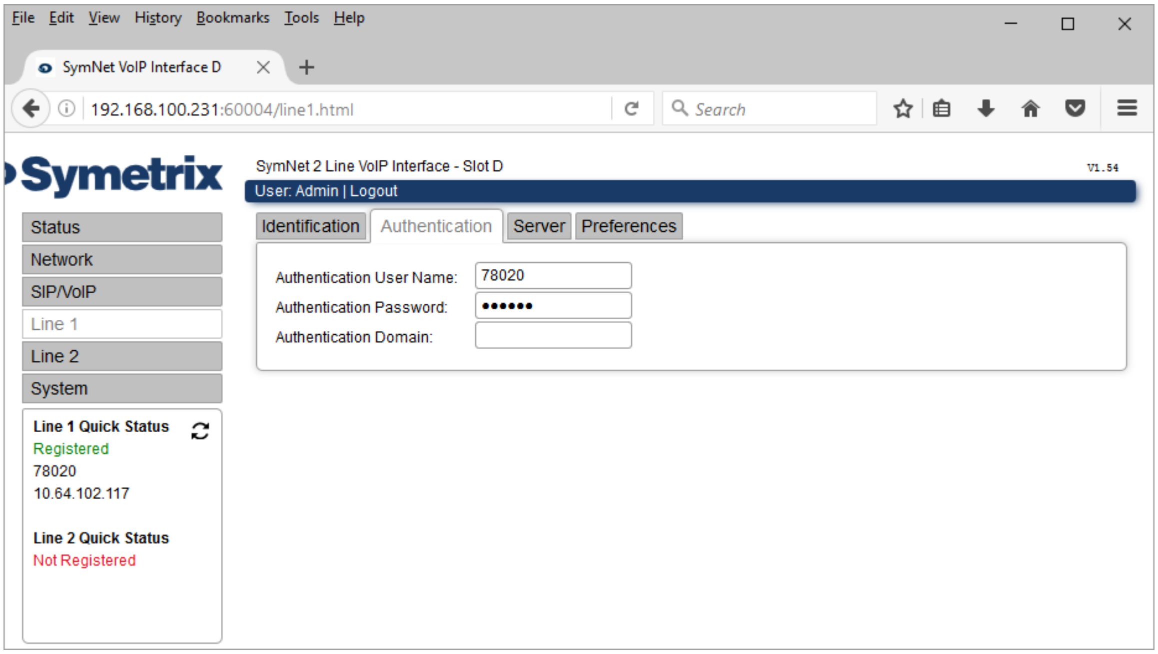


7.3.2. Configure SIP Line

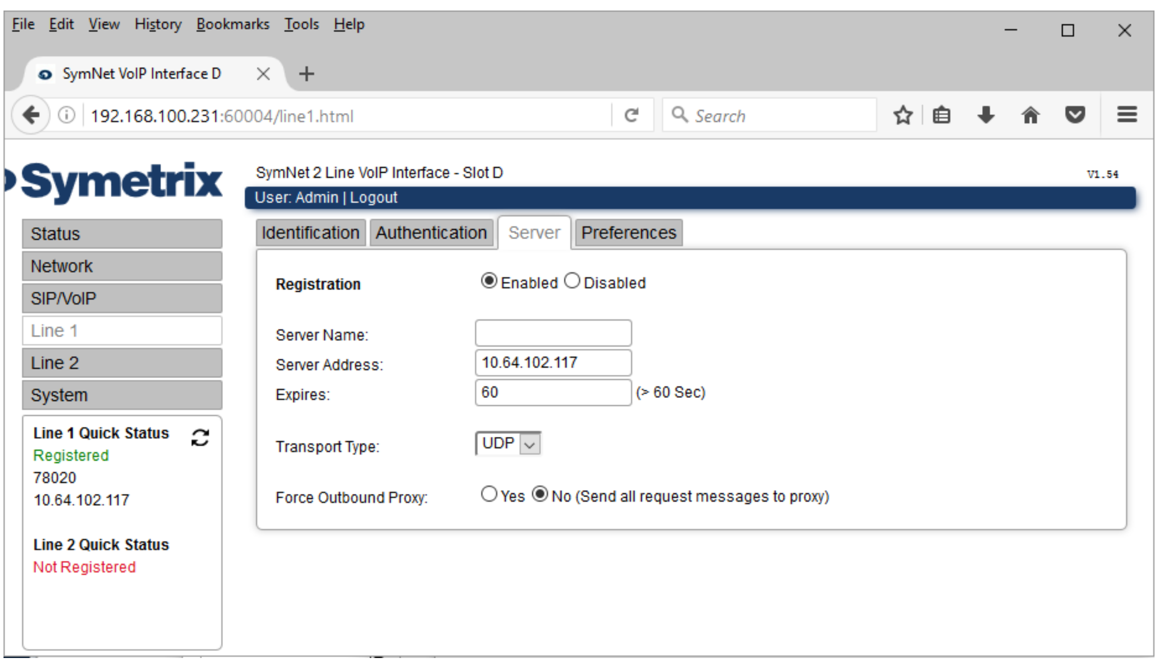
Select the **Line 1** option from the left pane. Under the Identification tab, set the **Display Name**, **User Name**, and **Local Phone Number** fields to the SIP extension (e.g., *78020*) assigned to the VoIP card in Session Manager, configured in **Section 6.7**. Click **Save** (not shown).



In the **Authentication** tab, set the **Authentication User Name** field to the SIP extension (e.g., *78020*) and set the **Authentication Password** to the password configured for the SIP user in **Section 6.7**. Click **Save** (not shown).

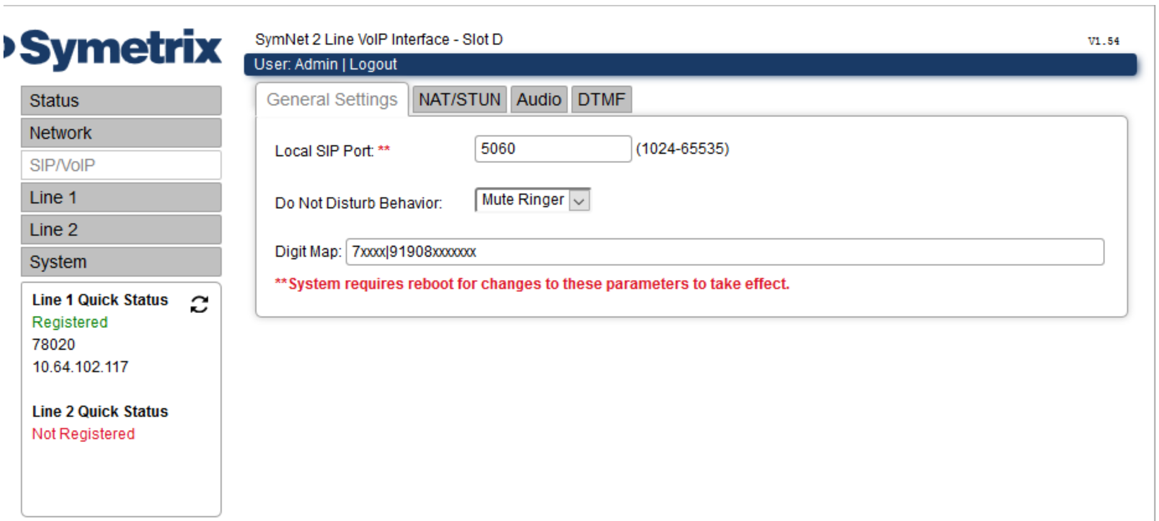


In the **Server** tab, set the **Server Address** to the IP address of the signaling interface of Session Manager (i.e., *10.64.102.117*). Specify the **Transport Type**, either *UDP* or *TCP*. Click **Save** (not shown).

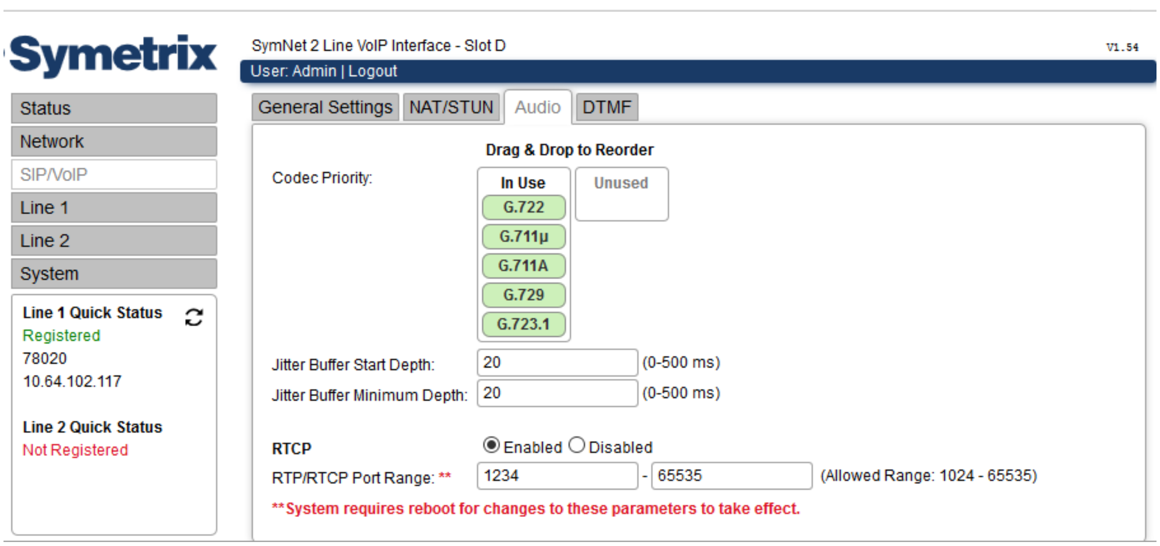


7.3.3. Configure SIP/VoIP Parameters

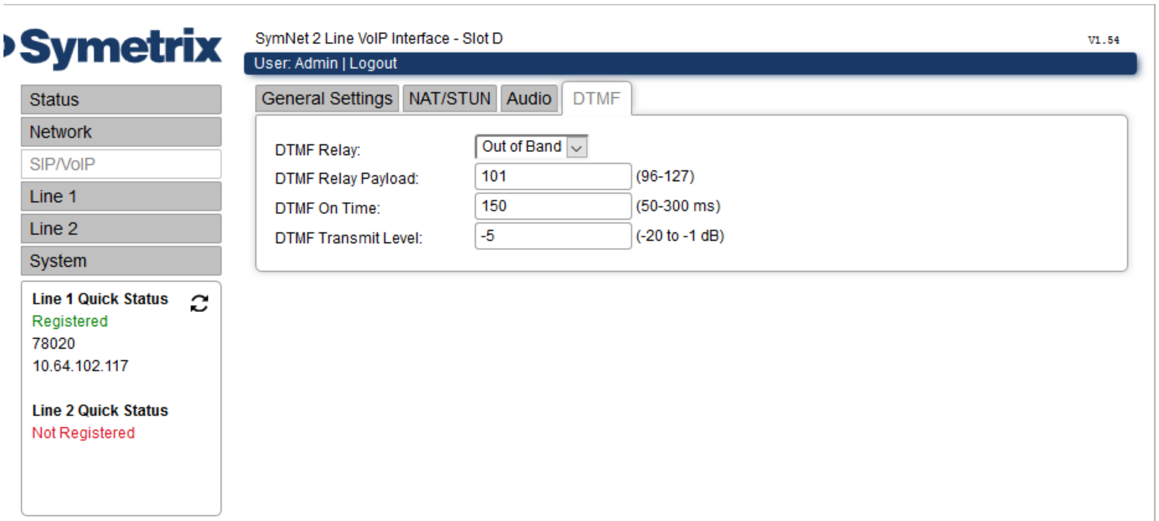
Select the **SIP/VoIP** option in the left pane. Under the **General Settings** tab, specify the **Digit Map**. For the compliance test, the VoIP card dialed local extensions that were 5-digits long beginning with “7” (i.e., *7xxxx*) and PSTN numbers that were 11-digits long in the 908 area code preceded with a “9” for the ARS feature access code (i.e., 91908xxxxxxxx). Therefore, the **Digit Map** field was set to *7xxxx | 91908xxxxxxx*. Click **Save** (not shown).



Under the **Audio** tab, specify the codecs to be supported. For the compliance test, G.711, G.729, and G.722 were used. Click **Save** (not shown).



Lastly, the default settings in the **DTMF** tab were used as shown below.



8. Verification Steps

This section provides the tests that may be performed to verify proper configuration of the Symetrix 2 Line VoIP Interface Card with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the Symetrix 2 Line VoIP Interface Card has successfully registered with Session Manager. In System Manager, navigate to **Elements → Session Manager → System Status → User Registrations** to check the registration status.

2. The SIP registration status of the 2 Line VoIP Interface Card can also be viewed in the VoIP Web Admin interface as shown in **Section 7.3**.

3. Verify basic telephony feature by establishing calls to the 2 Line VoIP Interface Card. Verify two-way audio, that the call can be placed on hold, and that a 3rd party can be joined into a conference.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Symetrix 2 Line VoIP Interface Card (installed in Symetrix Radius AEC) with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Symetrix 2 Line VoIP Interface Card successfully registered with Session Manager and basic and telephony features were verified. All test cases passed with observations noted in **Section 2.2.**

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1, Issue 2, May 2016, Document Number 03-300509.
[2] *Administering Avaya Aura® Session Manager*, Release 7.0.1, Issue 2, May 2016.

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.