



DSP-1282 & DSP-1283
Crestron Avia™ DSP (Secure)
with Avaya Aura® 7.1 Platform

Configuration Guide
Crestron Electronics, Inc.

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DSP-1282 & DSP-1283: Secure SIP Endpoint with Avaya Aura® 7.1 Platform

Introduction

This configuration guide describes the procedures required to configure Crestron Avia™ Digital Signal Processor (DSP) devices in a secure mode. The devices operate on the Avaya Aura® Communications Manager as Session Initiation Protocol (SIP) endpoints.

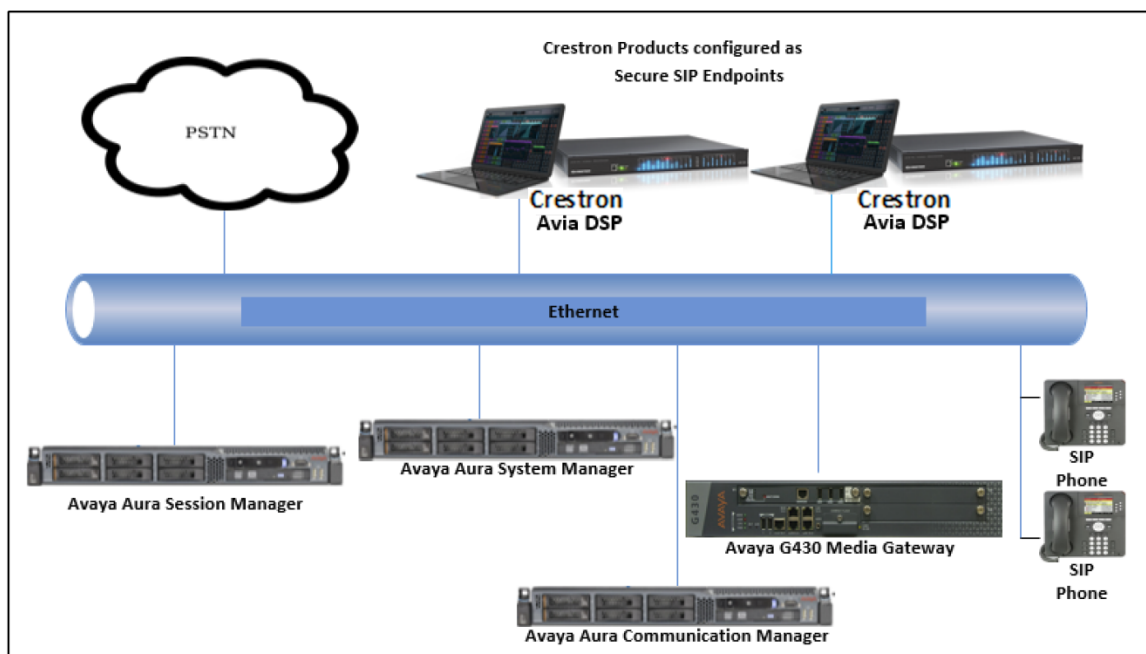
Audience

The intended audience includes those attempting to configure and use Crestron Avia DSP devices as secure SIP endpoints registered to Avaya Aura Communication Manager 7.1.

Topology

The diagram below shows the network topology for integration of a Crestron Avia DSP endpoint with Avaya Aura.

Secure SIP Endpoint Integration with Avaya Aura - Reference Network



The lab network consists of the following components:

- Avaya Aura Communication Manager
- Avaya Aura Session Manager
- Avaya Aura System Manager
- Avaya® SIP phones
- Avaya G430 Media Gateway
- Crestron Avia DSP as SIP endpoints

Software Requirements

- Avaya Aura Communication Manager v7.1.2.0.0.532.24184
- Avaya Aura Communication Manager Messaging v7.0.0.1.441.1
- Avaya Aura System Manager v7.1.2.0.057353
- Avaya Aura Session Manager v7.1.2.0.712004
- Avaya g430 Media Gateway v39.5.0/2
- Crestron Avia DSP-128 v 1.00.262.005

Hardware Requirements

- Cisco UCS-C240-M3S VMWare Host running ESXi 5.5
- Avaya Components either in a virtual environment or with separate hardware servers:
 - Avaya Aura Communication Manager
 - Avaya Aura Session Manager
 - Avaya Aura System Manager
 - Avaya G430 Media Gateway
 - Avaya Aura Communication Manager Messaging
- Public Switched Telephone Network (PSTN) gateway (Cisco 3845)
- Avaya Phones (2) in SIP
- Crestron Avia DSP devices (2):
 - Microphones for the DSP (2)
 - Speakers for the DSP (2)
 - Amplifiers for the DSP (2)
 - Appropriate cables for the above

Product Description

The Crestron Avia DSP products (DSP-1282 and DSP-1283, specifically) consist of a family of programmable digital audio signal processors intended for the commercial sound market. Each version provides 12 analog mic/line inputs and eight analog line outputs. The devices include a Local Area Network (LAN) connection and a Universal Serial Bus (USB) connection for programming and control. The programmable signal flow is a fixed topology with user-configurable input and output processing chains using a library of preset signal-specific DSP blocks.

Use the Crestron Avia tool to:

- Discover the device on the network
- Configure the SIP parameters
- Configure the mixers to allow 2-way communication on a SIP call

Save the audio configuration along with the SIP configuration as a project file. The project file can be loaded onto all of the DSPs that receive similar settings on a given project. Minor modifications may be necessary.

Use the Crestron Toolbox™ software to discover and control all Crestron devices on the network.

During the integration test, Crestron Toolbox can:

- Discover devices on the network
- Console connect to the devices
- Configure the Ethernet settings
- Upgrade firmware

Summary

This document describes how to configure the Crestron Avia DSP devices, in secure mode, as basic SIP endpoints. It also provides information on how to register devices to the Avaya Aura Session Manager with digest authentication.

Supported features include:

- Registration with digest authentication
- Basic calls with G711u and G711a codecs
- Dual-Tone Multi-Frequency (DTMF) support
- Early media support
- Retrieval of a parked call
- Transferee in a call transfer
- Conference participant
- Member of hunt group

Unsupported features include:

- Calls with non-secure (Real-time Transport Protocol (RTP) only) devices
- Caller ID presentation
- Call hold and resume
- Call forwarding on the device (forwarding can be configured on the Private Branch Exchange (PBX) for the Domain Name (DN) assigned to the endpoint)
- Call waiting
- Conference
- Attended call transfer
- Early attended call transfer
- Blind call transfer
- Shared line (configuration of shared line on device)
- Initiating call park
- Voice mail access and interaction

Known issues and limitations include:

- No support for caller ID on the Crestron Avia DSP.
- No support for MWI on the Crestron Avia DSP.
- The DSP does not support Music on Hold when integrated with the Avaya Aura PBX.
- The DSP does not support changes to DNS management when configured without DHCP settings via Toolbox.
- Intermittent issue with DTMF sent from Crestron Avia DSP to Avaya Media Gateway. The far end indicates duplicate and missing DTMF events.
- During a call from a secure DSP to the Avaya Communication Manager Messaging (CMM), there is incompatibility in Real-time Transport Control Protocol (RTCP) support between devices. The Avaya CMM only supports RTCP and does not support Secure Real-time Transport Control Protocol (SRTCP, sent by the DSP in secure mode). The Avaya CMM rejects SRTCP calls with a 488 "Not Acceptable Here" message.

Crestron Avia DSP Configuration

This section provides the following details:

- How to set up connections to the amplifier and speaker
- How to access the DSP on the network (once powered)
- How to configure the DSP for registration and integration with the PBX

Connections

Make the following connections:

- Connect microphone to DSP MIC/LINE INPUTS port 1
- Connect DSP LINE OUTPUTS port 1 to "Audio In" on amplifier
- Connect "Audio Out" of amplifier to speaker
- Connect LAN port to network

Device Discovery/Access

Use the Crestron Toolbox and the Crestron Avia tool to discover and access the connected LAN and/or VOIP ports) DSP devices.

Use the Help menu to assist when performing the discovery and configuration procedure.

Set Up SIP Interface

The DSP units have separate network interfaces for Voice over Internet Protocol (VoIP) and LAN on the rear panel. Configure either one for SIP calling. The default configuration binds SIP calling to the LAN interface. An optional console command binds the SIP interface to the VoIP connector. Configure all VoIP connections on a separate Virtual Local Area Network (VLAN) or subnet. VoIP connections cannot be on the same subnet as the LAN connection.

Ethernet

Use the **Ethernet** command to turn the VoIP port on/off.

```
DSP-1281>Ethernet ?
ETHERNET [<device_num> ON | OFF [/now]]
Device_num - 0 n
ON - enables VoI
OFF - disables VoIP
/now - take effect without a reboot
No parameter - displays the current setting
```

The VoIP port is off by default. The LAN port is not selectable.

```
<device_num> = 0 selects the LAN port
<device_num> = 1 selects the VoIP port
```

SIP Interface

Use the **sipinterface** command to bind all SIP activity, data, and traffic to the selected port. If a VLAN or exclusive VoIP network is available, bind to the VoIP port (recommended).

```
DSP-1281>sipinterface ?
Get or Set SIP Interface
SIPINTERFACE [LAN | VOIP]
LAN - normal LAN port
VOIP - VOIP port
No Parameter - Displays current setting
```

Set Up Routes

If the configured VoIP port is the SIP interface, add a static route to ensure that all SIP routing is via the VoIP port.

The following console commands (**routeadd**, **routedel**, **routeprint**, and **rutetrace**) support the static IP routing configuration:

```
DSP-1282>routeadd ?
ROUTEADD <destination> <netmask> <gateway> [/FORCE]
destination - destination IP address in dot decimal notation
netmask - netmask in dot decimal notation
gateway - gateway in dot decimal notation
/FORCE - force to add/delete even if failed to persist to NVRAM
```

```
DSP-1282>routedel ?
ROUTEDELETE <destination> <netmask> <gateway> [/FORCE]} | </ALL>
destination - destination IP address in dot decimal notation
netmask - netmask in dot decimal notation
gateway - gateway in dot decimal notation
/FORCE - force to add/delete even if failed to persist to NVRAM
/ALL - delete all routes from NVRAM
```

```
DSP-1282>routeprint ?
ROUTEPRINT - shows current routes
```

```
DSP-1282>routetrace ?
ROUTETRACE <IPaddress>
IPaddress - IP address in dot decimal notation
```

Device Configuration

The basic setup for a phone call requires:

- An analog input (such as from a microphone) routed out through the phone line
- Audio coming in from the phone line routed to an analog output (such as to an amplifier or speaker)

Configure the DSP Device

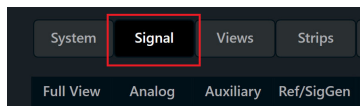
Use the Crestron Avia tool to select and configure the DSP device.

Input Configuration

To configure the analog input:

1. Click **Signal**.

Crestron Avia tool: Audio Input Configuration (1/4)



2. Under **Analog In 1** (first row), double click **Gain**. In the new window set the following:
 - a. Click **Mute** to **Off**.
 - b. Select **33** for the **Analog Gain**.
 - c. If a condenser microphone is being used, click **+48V** (phantom power) to **On**.

Crestron Avia Tool: Audio Input Configuration (2/4)



3. Under **Analog In 1** (first row), click **Ref/Phone Out** (right-most column) and enter **0** as the decibel value.

Crestron Avia Tool: Audio Input Configuration (3/4)



4. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
 - a. Move the **Send Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Input Configuration (4/4)

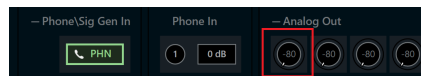


Output Configuration

To configure the analog output:

1. Under **Phone In 1** (first row), click **Analog Out** (left-most column) and enter **0** as the decibel value.

Crestron Avia Tool: Audio Output Configuration (1/3)



2. Under **Analog Out 1**, double click **LVL**. In the new window set the following:
 - a. Move the **Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Output Configuration (2/3)



3. Under **Phone\Sig Gen In**, click **PHN**. In the new window set the following:
 - a. Move the **Receive Level** slider to **0 db**.
 - b. Click **Mute** to **Off**.

Crestron Avia Tool: Audio Output Configuration (3/3)



Configure the SIP Parameters

From the open **PHN - In 1 - Phone Receive** window, select and configure the SIP parameters.

1. With **SIP** selected, click the chevron at the right top corner to expand the window.
Crestron Avia Tool: Phone Dialer, SIP Parameters Configuration



2. Enter the extension configured on Avaya Aura CM for the **Local Extension** for this device. This example uses **6625**.
3. Enter the Avaya Aura Session Manager PBX for the **SIP Server IP Address**. This example uses **10.89.26.7**.
4. Enter the SIP server port (**5061**) for the **Port**.
5. Enter the same end user name configured for the Avaya Aura Session Manager with the digest authentication credentials for the **SIP Server User Name**.
6. Enter the same password as configured for the Avaya Aura Session Manager end user digest credentials for the **SIP Server Password**.

Certificates

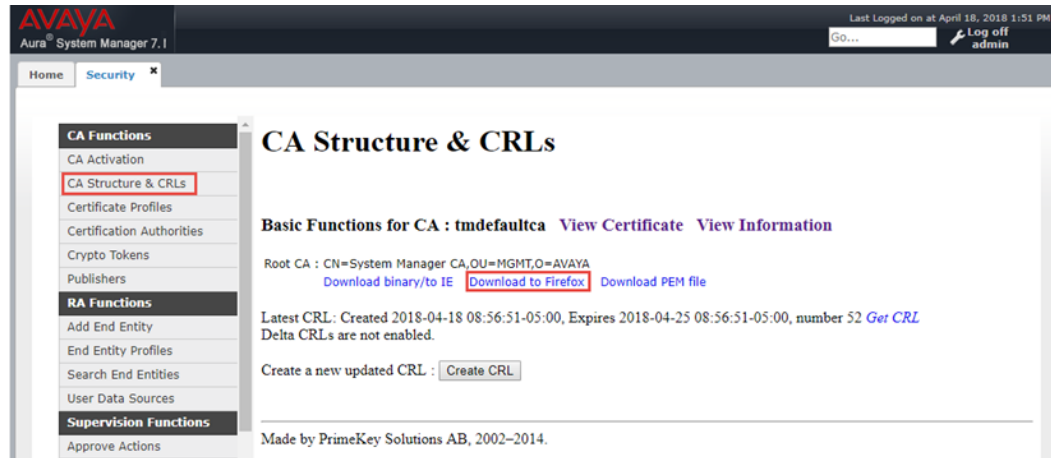
For a successful TLS handshake between the DSP device and the Avaya Aura, add the following root certificate to the DSP: A rootCA certificate (systemmanagerca.cer).

Download this certificate from the Avaya System Manager. The DSP requires the certificate to validate the Avaya Aura when the "Enable Server Validation" is enabled.

To download Avaya Aura CA from Avaya Aura System Manager:

1. Click **Home > Services > Security**.

Download Root Certificate from Avaya System Manager



2. Click **CA Structure & CRLs**.
3. From the CA Structure & CRLs window, click on **Download to Firefox** (for this example).
4. Save the file as **systemmanagerca.cer** (for this example).

Copy Certificates

Copy the Root Certificate into the DSP device under the directory /user/cert **by doing SFTP**.

Import and Assign Root Certificate

Use the Crestron Toolbox text console utility (in the Crestron Avia DSP console) to import and assign the root certificate.

1. Type the command `certificate addf systemmanagerca.cer root` (where `RootCA` is the name of the root certificate uploaded in the previous section).

Root Certificate - Successful Import

```
DSP-1282>certificate addf systemmanagerca.cer root
Import successful.
```

2. Verify the uploaded root certificate and note the # of the certificate.

List of Trusted Root Certificates in the Device

```
DSP-1282>Console command returned error.
DSP-1282>siptrustedcas lista
TableStart: [Sip Trusted Certification Authorities]
-----|-----|-----|-----|
# | Name | UID | In Use |
-----|-----|-----|-----|
145 | EC-ACC | -11D4C2142BDE21EB579D53FB0C223BFF | No |
146 | thawte Primary Root CA | 344ED55720D5EDEC49F42FCE37DB2B6D | No |
147 | Visa eCommerce Root | 1386354D1D3F06F2C1F96505D5901C62 | No |
148 | skypalabsi-NC01-CA | 343F63A0387235A1449ABB5676612403 | No |
149 | System Manager CA | 0853D8A9BB8E1C09 | No |
-----|-----|-----|-----|
```

Root Certificate - Assignment

```
DSP-1282>siptrustedcas use 149
Sip Trusted Certification Authorities USING System Manager CA |0853D8A9BB8E1C09
DSP-1282>
DSP-1282>siptrustedcas listu
TableStart: [Sip Trusted Certification Authorities]
-----|-----|-----|-----|
# | Name | UID | In Use |
-----|-----|-----|-----|
149 | System Manager CA | 0853D8A9BB8E1C09 | Yes |
-----|-----|-----|-----|
```

Avaya Aura Communication Manager Configuration

This section describes the Avaya Aura Communication Manager (Avaya CM) configuration necessary to integrate the Crestron Avia DSP devices in secure mode.

NOTE: Confirm that the general installation and basic Avaya CM configuration have been administered.

Node Names

Configure the node IP for Avaya Aura Session Manager (SM) and Avaya CM.

Use the **change name-names ip** command to add the node name. This example adds **ASM1** and **procr** with their respective IPs.

- Use **ASM1**, an Avaya Aura Session Manager, to register the SIP phones and third-party SIP devices.
- User **procr** to register SIP trunk between Avaya CM and Avaya SM.

Avaya Aura CM: Configure Node

```
display node-names ip
IP NODE NAMES
  Name          IP Address
ASM1           10.89.26.7
default        0.0.0.0
procr          10.89.26.4
procr6         ::

( 4 of 4 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
Command:
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```


Dial Plan Analysis

Configure several dial strings to ensure complete test coverage. This example includes calling between stations, calling to PSTN, and accessing PBX features.

Use the **change dialplan analysis** command to configure the following dial patterns for this example:

1. Enter **5** for the mail number **Dialed string**.
2. Enter **6** for the station number **Dialed string**.
3. Enter **8** for the feature access code **Dialed string**.
4. Enter **9** for the feature access code **Dialed string**.
5. Enter ***** for the feature access code **Dialed string**.
6. Enter **#** for the dial access code **Dialed string**.

Use the **display dialplan analysis** command to view the configured dial strings/codes.

Avaya Aura CM: Dial Plan Analysis

```
display dialplan analysis                                     Page 1 of 12
DIAL PLAN ANALYSIS TABLE
Location: all                                             Percent Full: 3

Dialed  Total  Call  Dialed  Total  Call  Dialed  Total  Call
String  Length Type  String  Length Type  String  Length Type
0       1      attd
1       4      ext
2       4      ext
5       4      ext
6       4      ext
7       4      ext
8       1      fac
9       1      fac
*       3      fac
#       4      dac

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

IP Network Region

The example configures all the SIP phones in ip-network-region-1. Configure the **Domain** name and **Codec Set** parameters.

Avaya Aura CM: ip-network-region

```
change ip-network-region 1                                     Page 1 of 20
IP NETWORK REGION
Region: 1             NR Group: 1
Location:             Authoritative Domain: lab.tekvizion.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: 65535
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
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

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the PBX and PSTN.

Avaya Aura CM: Codec Configuration

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

IP MEDIA PARAMETERS

Codec Set: 1

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

Media Encryption      Encrypted SRTCP: best-effort
1: 1-srtp-aescm128-hmac80
2: 7-srtp-aescm128-hmac80-unenc-unauth
3:
4:
5:

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

This example uses 1 for the **Codec Set**. The Crestron DSP device supports and includes G.711A and G.711MU in this set. To test with the DSP, enter **G.711A** and **G.711MU** in the **Audio Codec** column of the table. Use default values for all other fields. Configure media encryption (SRTP and SRTCP). The **Media Encryption** section shows sample values.

Signaling Group

This example configures two signaling groups.

- Signaling Group 3
This group supports communication between SM and CM for SIP phone registration and features.
- Signaling Group 10
This group supports PSTN calling on ISDN-PRI.

Use the **add signaling group n** command to create a signaling group system (where **n** represents the signaling group number for this example).

Avaya Aura CM: Signaling Group Configuration for Phones

```
display signaling-group 3                               Page 1 of 2
SIGNALING GROUP
Group Number: 3                                     Group Type: sip
IMS Enabled? n                                     Transport Method: tls
Q-SIP? n
IP Video? n                                       Enforce SIPS URI for SRTP? n
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: ASM7
Near-end Listen Port: 5061                         Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain: lab.tekvizion.com
Incoming Dialog Loopbacks: eliminate               Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                          RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3                 Direct IP-IP Audio Connections? n
Enable Layer 3 Test? y                             IP Audio Hairpinning? n
Alternate Route Timer(sec): 6
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

To configure Signaling Group 3 (for this example):

1. Enter **3** for the **Group Number**.
2. Enter **sip** for the **Group Type**.
3. Enter **tls** for the **Transport Method**.
4. Enter **SM** for the **Peer Server**.
5. Enter **procr** for the **Near-end Node Name**.
6. Enter **5061** for the **Near-end Listen Port**.
7. Enter **ASM7** for the **Far-end Node Name**.
8. Enter **5061** for the **Far-end Listen Port**.
9. Enter **1** for the **Far-end Network Region**.
10. Enter **lab.tekvizion.com** for the **Far-end Domain**.
11. Enter **n** for **Direct IP-IP Audio Connections**.

Avaya Aura CM: Signaling Group Configuration for PSTN

```
change signaling-group 10                               Page 1 of 5
                SIGNALING GROUP
Group Number: 10   Group Type: isdn-pri
Associated Signaling? y   Max number of NCA TSC: 0
Primary D-Channel: 001V224   Max number of CA TSC: 0
Trunk Group for NCA TSC:       
Trunk Group for Channel Selection: 10   X-Mobility/Wireless Type: NONE
TSC Supplementary Service Protocol: a   Network Call Transfer? n

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

To configure Signaling Group 10 (for this example):

1. Enter **10** for the **Group Number**.
2. Enter **isdn-pri** for the **Group Type**.
3. Enter **001V224** for the **Primary D-Channel**.

Trunk Groups

Configure two trunk groups (for this example):

- Trunk Group 3
This group accesses the stations registered to the Avaya Session Manager.
- Trunk Group 10
This group sends a 10/11-digit calling number to PRI trunk or PSTN.

Use the **add trunk group n** command to add a new trunk group (where **n** represents the trunk group number).

Trunk Group 3

Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (1/4)

```
change trunk-group 3                                     Page 1 of 21
TRUNK GROUP
Group Number: 3                                         Group Type: sip   CDR Reports: y
Group Name: SIP Phone                                  COR: 1           TN: 1           TAC: #003
Direction: two-way                                     Outgoing Display? n
Dial Access? n                                         Night Service:
Queue Length: 0
Service Type: tie                                       Auth Code? n
Member Assignment Method: auto
Signal Group: 3
Number of Members: 10
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

Configure Trunk Group 3 (entries for steps 1 through 7 are for this example):

1. Enter **3** for the **Group Number**.
2. Enter **SIP Phone** for the **Group Name**.
3. Enter **sip** for the **Group Type**.
4. Enter **tie** for the **Service Type**.
5. Enter **#003** for the **TAC**.
6. Enter **3** for the **Signaling Group**.
7. Enter **10** for the **Number of Members**.

Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (2/4)

```
change trunk-group 3                                     Page 2 of 21
  Group Type: sip

TRUNK PARAMETERS

  Unicode Name: auto

                                         Redirect On OPTIM Failure: 5000

  SCCAN? n                                         Digital Loss Group: 18
                                         Preferred Minimum Session Refresh Interval(sec): 1800
  Disconnect Supervision - In? y Out? y

  XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n

  Caller ID for Service Link Call to H.323 1xC: station-extension

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

8. Enter 1800 for the Preferred Minimum Session Refresh Interval (sec.).

Avaya Aura CM: Trunk Group to Session Manager - TrunkGroup 3 (3/4)

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

  Suppress # Outpulsing? n Numbering Format: private
                                         UI 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
```

9. Enter private for the Numbering Format.

Avaya Aura CM: Trunk Group to Session Manager - Trunk Group 3 (4/4)

```
display trunk-group 3                                     Page 4 of 21
PROTOCOL VARIATIONS
    Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
    Send Transferring Party Information? n
    Network Call Redirection? n
    Send Diversion Header? n
    Support Request History? y
    Telephone Event Payload Type: 96
    Convert 180 to 183 for Early Media? n
    Always Use re-INVITE for Display Updates? y
    Identity for Calling Party Display: P-Asserted-Identity
    Block Sending Calling Party Location in INVITE? n
    Accept Redirect to Blank User Destination? n
    Enable Q-SIP? n
    Interworking of ISDN Clearing with In-Band Tones: keep-channel-active
    Request URI Contents: may-have-extra-digits
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```


Trunk Group 10

Avaya Aura CM: Trunk Group to PRI/PSTN - Trunk Group 10

```
display trunk-group 10                                     Page 1 of 21
TRUNK GROUP
Group Number: 10                                         Group Type: isdn          CDR Reports: y
Group Name: OUTSIDE CALL                                COR: 1                   TN: 1          TAC: #010
Direction: two-way                                     Outgoing Display? n     Carrier Medium: PRI/BRI
Dial Access? n                                         Busy Threshold: 255     Night Service:
Queue Length: 0
Service Type: public-ntwrk                             Auth Code? n           TestCall ITC: rest
Far End Test Line No:
TestCall BCC: 4
F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
```

Configure Trunk Group 10 (for this example):

1. Enter **10** for the **Group Number**.
2. Enter **OUTSIDE CALL** for the **Group Name**.
3. Enter **isdn** for the **Group Type**.
4. Enter **public-ntwrk** for the **Service Type**.
5. Enter **#010** for the **TAC**.

Route Pattern

The route pattern defines which trunk group will be used for the call and performs any necessary digit manipulation. Use the **change route pattern n** command (where **n** represents the route pattern number to configure the parameters for the PSTN trunk route pattern).

Use Route Pattern 3 for calling extensions via Avaya Aura Session Manager.

Avaya Aura CM: Route Pattern for SIP Phones

```

display route-pattern 3                                     Page 1 of 3
Pattern Number: 3      Pattern Name: SIP Phone
SCCAN? n      Secure SIP? n      Used for SIP stations? y
Primary SM: ASM7      Secondary SM:
Grp FRL NPA Pfx Hop Toll No.  Inserted      DCS/ IXC
No           Mrk Lmt List Del  Digits      QSIG
                                           Intw
1: 3      0
2:
3:
4:
5:
6:
                                           n  user
                                           n  user
                                           n  user
                                           n  user
                                           n  user
                                           n  user

      BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM Sub  Numbering LAR
      0 1 2 M 4 W      Request      rest
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

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
  
```

Use Route Pattern 10 for calling PSTN.

Avaya Aura CM: Route Pattern for PSTN (PRI)

```

display route-pattern 10                                   Page 1 of 3
Pattern Number: 10     Pattern Name: PRI
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
                                           Intw
1: 10     0
2:
3:
4:
5:
6:
                                           n  user
                                           n  user
                                           n  user
                                           n  user
                                           n  user
                                           n  user

      BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM Sub  Numbering LAR
      0 1 2 M 4 W      Request      rest
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

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg
  
```

Auto Alternative Routing

Use the **change aar analysis n** command (where **n** represents the first digit of the extension numbers for making calls).

Avaya Aura CM: Auto Alternative Routing Analysis 6

```
display aar analysis 6
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE
Location: all Percent Full: 3

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Recd
	Min	Max				
6	4	4	3	unku	n	

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Automatic Route Selection

Use the `change ars analysis n` command (where `n` represents the pattern for making PSTN calls).

Avaya Aura CM: Auto Routing Selection Analysis

```
display ars analysis 2
```

Page 1 of 2

ARS DIGIT ANALYSIS TABLE
Location: all Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
214	10	10	10	natl		n

F1=Cancel F2=Refresh F3=Submit F4=Clr Fld F5=Help F6=Update F7=Nxt Pg F8=Prv Pg

Avaya Aura Session Manager

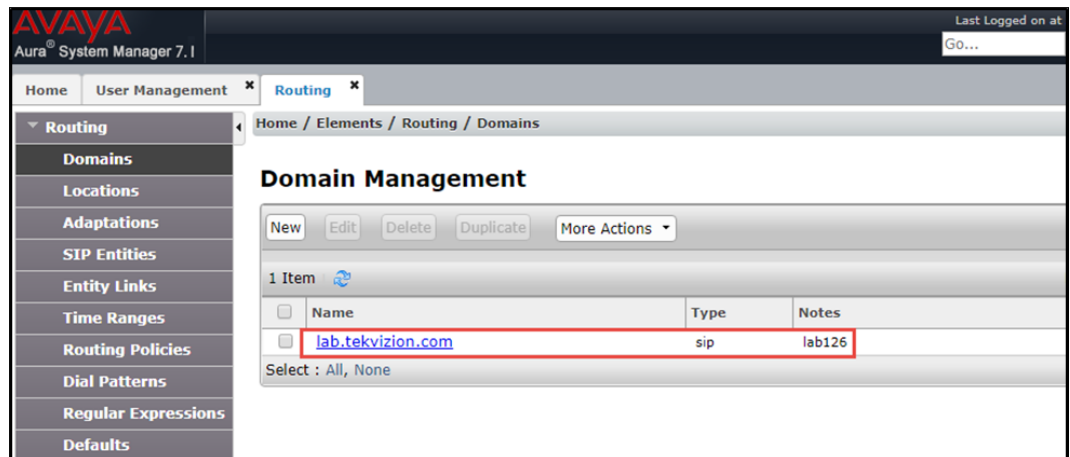
Domain

To route calls, create a SIP domain for each domain administered by the Session Manager.

To configure a domain:

1. Click **Home** > **Routing** > **Domains**.
2. Click **New**.

Avaya Aura SM: Domain Configuration



The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes 'Home', 'User Management', and 'Routing'. The left sidebar lists various configuration options, with 'Domains' selected under the 'Routing' section. The main content area is titled 'Domain Management' and features a table with the following data:

Name	Type	Notes
lab.tekvizion.com	sip	lab126

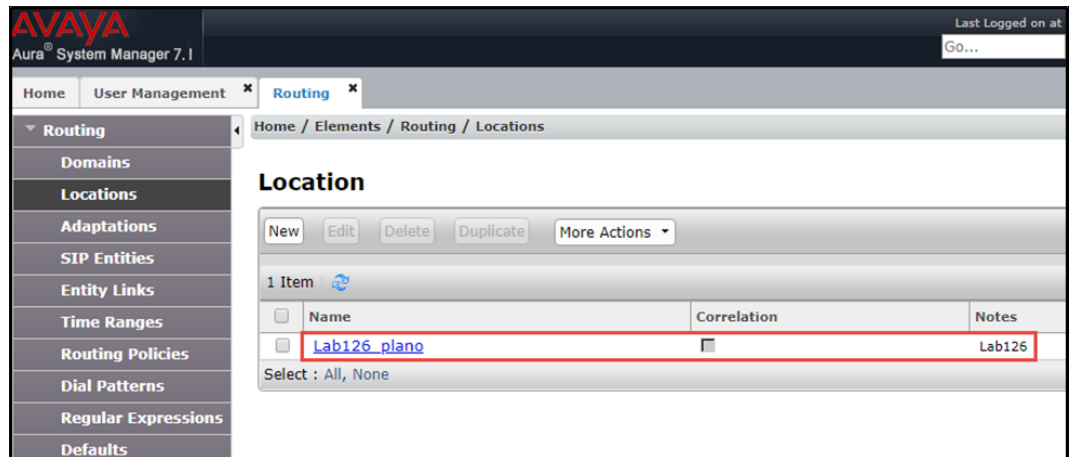
3. Enter the domain name for the **Name**. This example uses **lab.tekvizion.com**.
4. Select **sip** for the **Type**.
5. Enter a brief description for the **Notes** (optional).
6. Click **Commit** to save (not shown).

Location

Use locations to identify logical and/or physical locations where SIP entities reside for the purposes of bandwidth management and call admission control.

To add a location:

1. Click **Routing > Locations**.
2. Click **New**.
Avaya Aura SM: Location Configuration

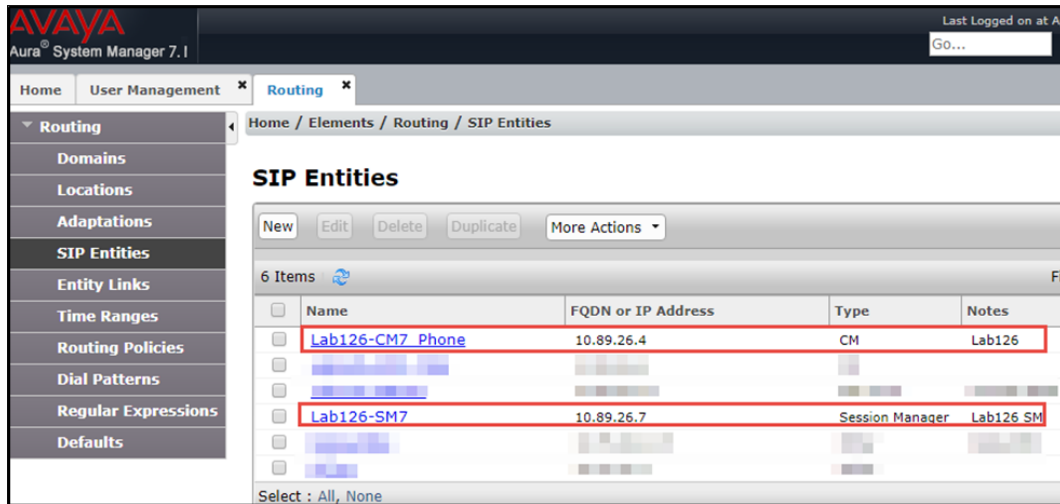


3. In the **General** section, do the following:
 - a. Enter a descriptive name for the **Name** of the location. This example uses **Lab126-plano**.
 - b. Enter a brief description for the **Notes** (optional).
 - c. Use the default values for all remaining fields.
4. Click **Commit** to save (not shown).

SIP Entity

Add a SIP entity for each SIP telephony system connected to the Session Manager, which includes Communication Manager and Avaya Communication Manager Messaging Component.

Avaya Aura SM: SIP Entity



The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes 'Home', 'User Management', and 'Routing'. The left sidebar lists various configuration options, with 'SIP Entities' selected. The main content area displays the 'SIP Entities' configuration page, which includes a table with 6 items. The table has columns for Name, FQDN or IP Address, Type, and Notes. Two rows are highlighted with red boxes: 'Lab126-CM7 Phone' (Type: CM, Notes: Lab126) and 'Lab126-SM7' (Type: Session Manager, Notes: Lab126 SM).

Name	FQDN or IP Address	Type	Notes
Lab126-CM7 Phone	10.89.26.4	CM	Lab126
Lab126-SM7	10.89.26.7	Session Manager	Lab126 SM

To add a SIP entity:

1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: Sip Entity - CM Configuration

The screenshot shows the Avaya Aura System Manager 7.1 interface. The main content area is titled "SIP Entity Details" and is divided into sections: "General", "SIP Timer B/F (in seconds)", "Loop Detection", and "Monitoring". The "General" section is highlighted with a red box and contains the following fields:

- Name:** Lab126-CM7
- FQDN or IP Address:** 10.89.26.4
- Type:** CM
- Notes:** (empty)
- Adaptation:** to_cm
- Location:** Lab126_plano
- Time Zone:** America/Chicago

Below the "General" section, the following fields are visible:

- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- Securable:**
- Call Detail Recording:** none
- Loop Detection:**
 - Loop Detection Mode:** On
 - Loop Count Threshold:** 5
 - Loop Detection Interval (in msec):** 200
- Monitoring:**
 - SIP Link Monitoring:** Use Session Manager Configuration
 - CRLF Keep Alive Monitoring:** Use Session Manager Configuration
 - Supports Call Admission Control:**
 - Shared Bandwidth Manager:**

The "Commit" and "Cancel" buttons are located at the top right of the form.

3. In the **General** section, do the following:
 - a. Enter a descriptive name for the **Name**. This example uses **Lab126-CM7** for the Avaya CM.
 - b. Enter the FQDN or IP address of the SIP entity interface used for SIP signaling for the **FQDN or IP Address**. This example uses **10.89.26.4**.
 - c. Select **Session Manager** (for Session Manager), **CM** (for Communication Manager), and **Other** (for the Avaya SBCe) for the **Type**.
 - d. Select **Lab126-plano** (a location previously defined) for the **Location**.
 - e. Select the time zone for the location in the previous step for the **Time Zone**.
 - f. Scroll to the **Port** section of the **SIP Entity Details** screen to define the ports used by Communication Manager. Click **Add** and enter the following values:
 - i. Enter the port number on which the CM listens for SIP requests for the **Port**. This example uses **5060**.
 - ii. Select the protocol used to send SIP requests for the **Protocol**. This example uses **TCP**.
 - iii. Use the default values for all remaining fields.

To add a SIP entity for the Avaya SM:

1. Click **Routing** > **SIP Entities**.
2. Click **New**.

Avaya Aura SM: SIP Entity - SM Configuration

The screenshot shows the Avaya Aura System Manager 7.1 interface. The main content area is titled "SIP Entity Details" and is divided into "General" and "Monitoring" sections. The "General" section contains several fields for configuring a SIP entity. A red box highlights the following fields and their values:

- Name:** Lab126-SM7
- FQDN or IP Address:** 10.89.26.7
- Type:** Session Manager
- Notes:** Lab126 SM
- Location:** Lab126_plano
- Outbound Proxy:** (empty)
- Time Zone:** America/Chicago
- Minimum TLS Version:** Use Global Setting

Below the "General" section, the "Monitoring" section includes:

- SIP Link Monitoring:** Use Session Manager Configuration
- CRLF Keep Alive Monitoring:** Use Session Manager Configuration

At the top right of the form area, there are "Commit" and "Cancel" buttons.

3. In the **General** section, do the following (for this example):
 - a. Enter **Lab126-SM7** for the **Name** (for a SIP entity of Avaya SM) .
 - b. Enter **10.89.26.7** for the **FQDN or IP Address**.
 - c. Select **Session Manager** for the **Type**.
 - d. Enter **Lab126 SM** for the **Notes**.
 - e. Select **DomainAdapter** for the **Adaptation**.
 - f. Select **Lab126-plano** for the **Location**.
 - g. Select **America/Chicago** for the **Time Zone**.

Entity Links

A SIP trunk between Avaya Session Manager and a telephony system is an entity link. This example creates an entity link.

To add Avaya CM as an entity link:

1. Click **Routing > Entity Links**.
2. Click **New**.

Avaya Aura SM: Avaya CM Entity Link Configuration

Entity Links

Override Port & Transport with DNS SRV:

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* Lab126-SM7_Lab126-CM	Lab126-SM7	TLS	* 5061	Lab126-CM7_Phone	* 5061	trusted	<input type="checkbox"/>

Select : All, None

3. In the new row that is displayed, do the following:
 - a. Enter a descriptive name for the **Name**.
 - b. Select the Session Manager for **SIP Entity 1**.
 - c. Select **TLS** (for the **Protocol** for this example).
 - d. Enter **5061** for the **Port** (for this example).
 - e. Select the Communication Manager for **SIP Entity 2**.
 - f. Enter **5061** for the **Port** (for this example).
 - g. Select **trusted** for the **Connection Policy**.
4. Click **Commit** to save.

User Configuration for Each Device/Phone

Configure a user for each phone and Crestron device:

1. Click **Home** > **User Management** > **Manage Users**.
2. Click **Add New**. The **User Profile View** window appears.

Avaya Aura SM: User Configuration (1/3)

The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes 'Home' and 'User Management'. The left sidebar lists various management options, with 'Manage Users' selected. The main content area displays the 'User Profile View' for the user '6625@lab.tekvizion.com'. The 'Identity' tab is active, showing a 'User Provisioning Rule' dropdown and an 'Identity' section. The 'Identity' section contains several fields: 'Last Name' (DSP1), 'Last Name (Latin Translation)' (DSP1), 'First Name' (Crestron1), 'First Name (Latin Translation)' (Crestron1), 'Middle Name', 'Description', 'Update Time' (March 26, 2018 11:27:07), 'Login Name' (6625@lab.tekvizion.com), 'Email Address', 'User Type' (Basic), 'Source' (local), 'Localized Display Name' (DSP1, Crestron1), 'Endpoint Display Name' (DSP1, Crestron1), 'Title', 'Language Preference' (English (United States)), and 'Time Zone'. Red boxes highlight the 'Last Name', 'First Name', and 'Login Name' fields.

3. Enter **DSP1** for the **Last Name** (for this example).
4. Enter **Crestron1** for the **First Name** (for this example).
5. Enter **6625@lab.tekvizion.com** for the **Login Name** (for this example).
6. Click the **Communication Profile** tab.

Avaya Aura SM: User Configuration (2/3)

The screenshot shows the Avaya Aura System Manager 7.1 interface for editing a user profile. The user is 6625@lab.tekvizion.com. The Communication Profile section includes a password field, a Name field set to 'Primary', and a Communication Address table. The Session Manager Profile section includes a SIP Registration field set to 'Lab126-SM7' and a table for session manager settings.

Type	Handle	Domain
Avaya SIP	6625	lab.tekvizion.com

Primary	Secondary	Maximum
9	0	9

7. Enter the desired SIP user registration password for the **Communication Profile Password**.
8. Confirm the password.
9. Scroll down to the **Communication Address** subsection and click **New** to add a new address.
10. Enter **Avaya SIP** for the **Type**.
11. Under **SIP Registration**, enter **Lab126-SM7** for the **Primary Session Manager**.

Avaya Aura SM: User Configuration (3/3)

The screenshot displays the Avaya Aura SM User Configuration interface. It is divided into three main sections: Call Routing Settings, Call History Settings, and CM Endpoint Profile. The CM Endpoint Profile section is highlighted with a red box and contains the following fields and options:

- Call Routing Settings:**
 - * Home Location: Lab126_plano
 - Conference Factory Set: (None)
- Call History Settings:**
 - Enable Centralized Call History?:
- CM Endpoint Profile:**
 - * System: Lab126-CM7
 - * Profile Type: Endpoint
 - Use Existing Endpoints:
 - * Extension: 6625 (with a link to Display Extension Ranges and an Endpoint Editor button)
 - Template: Select/Reset
 - Set Type: 9600SIP
 - Security Code: [Empty]
 - Port: S00002
 - Voice Mail Number: [Empty]
 - Preferred Handle: (None)
 - Calculate Route Pattern:
 - Sip Trunk: aar
 - Enhanced Callr-Info display for 1-line phones:
 - Delete Endpoint on Unassign of Endpoint from User or on Delete User:
 - Override Endpoint Name and Localized Name:
 - Allow H.323 and SIP Endpoint Dual Registration:

12. Check **CM Endpoint Profile**.
13. Select **Lab126-CM7** for the **System** (for this example).
14. Select **Endpoint** for the **Profile Type** (for this example).
15. Enter **6625** for the **Extension** (for this example).
16. Click **Done**.

Crestron Electronics, Inc.
15 Volvo Drive, Rockleigh, NJ 07647
Tel: 888.CRESTRON
Fax: 201.767.7656
www.crestron.com



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10.18
Specifications subject to
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