



Application Notes for Talkphone VOIP-200 Series IP Call Stations 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 Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Talkphone VOIP-200 Series IP Call Stations registered with Avaya Aura® Session Manager via SIP.

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 Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Talkphone VOIP-200 Series IP Call Stations registered with Session Manager via SIP.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkphone VOIP-200 Series IP Call Stations, Avaya H,323, SIP, Digital, Analog telephones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the Avaya IP phones. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Talkphone VOIP-200 Series IP Call Stations come back into service after re-connecting the Ethernet cable or rebooting the IP Call Station.

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

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

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

For the testing associated with this Application Note, the interface between Avaya systems and the Talkphone VOIP 200 did not include use of any specific encryption features.

Encryption (TLS/SRTP) was used internal to the enterprise between Avaya products.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Talkphone IP Call Station with Session Manager.
- Inbound and outbound calls between Talkphone IP Call Station and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled.
- Inbound and outbound calls between the Talkphone IP Call Station and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the Avaya IP phone.
- Use of number lists on the Talkphone IP Call Station.
- Proper system recovery after a restart of the Talkphone IP Call Station and loss of IP connectivity.

2.2. Test Results

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

On-hold call that is established between Talkphone VOIP 200 and Avaya H.323 endpoint gets dropped when Avaya H.323 endpoint resumes the call. The issue only happens with direct media enabled and with Avaya H.323 endpoint. If the destination of the call originated by the Talkphone VOIP 200 may be an Avaya H.323 endpoint, it is recommended to disable direct media on the Talkphone VOIP 200 station.

2.3. Support

For technical support and information on Talkphone VOIP-200 Series IP Call Stations, contact Talkphone support at:

Address : 7530 North Natchez Ave.
Niles, IL 60714
Telephone : (773) 539-1100
Fax : (773) 539-1241
Email : info@talkphone.com
Web : www.talkphone.com

3. Reference Configuration

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

- Avaya Aura® System Manager, Avaya Aura® Session Manager, Avaya Aura® Communication Manager and Avaya Aura® Media Server running on Virtualized Environment.
- Avaya Aura Messaging has SIP trunk connected to Session Manager and used as Voicemail system for the endpoint.
- Avaya G450 Media Gateway registers to Communication Manager and has PRI trunk to simulated PSTN.
- Session Manager has SIP trunk to simulated PSTN
- Avaya 96x1 H323 and SIP Deskphones were used to place and receive call to/from Talkphone VOIP station.
- Talkphone VOIP-200 IP Call Stations registered with Avaya Aura® Session Manager.

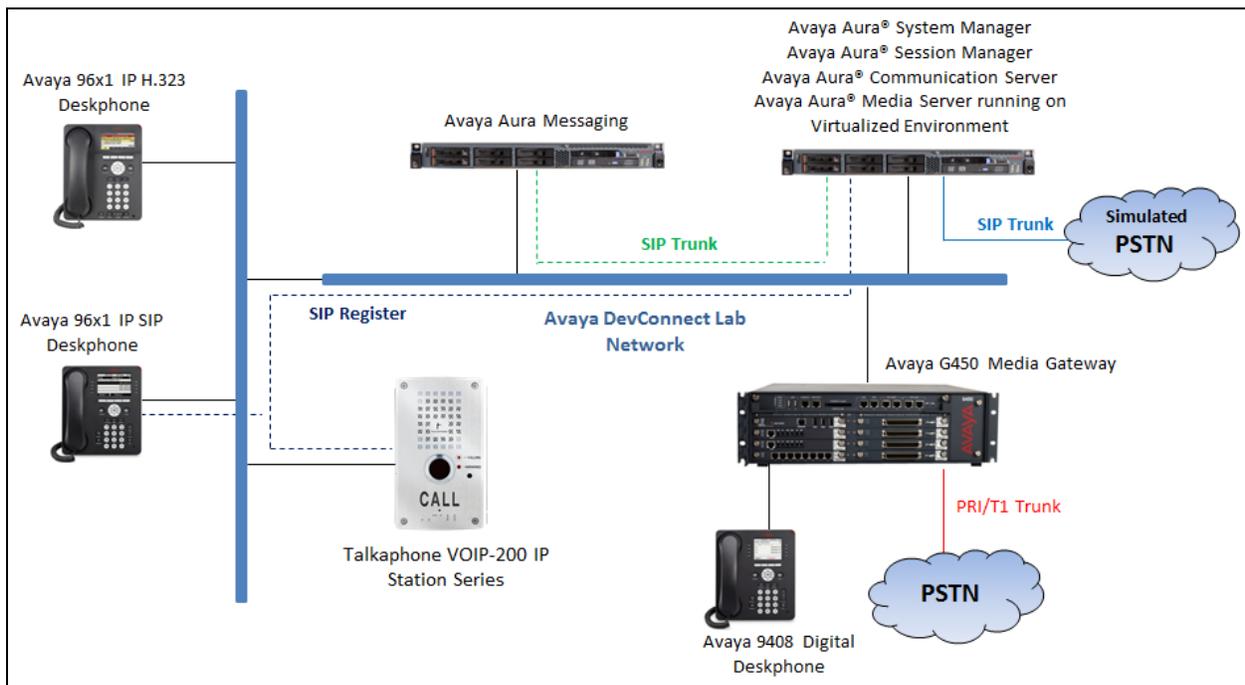


Figure 1: Avaya SIP Network with Talkphone VOIP-200 Series IP Call Stations

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Virtualized Environment	R017x.01.0.532.0 7.1.1.0.0.532.23985
Avaya Aura® System Manager running on Virtualized Environment	7.1.1.0.046931
Avaya Aura® Session Manager running on Virtualized Environment	7.1.1.0.711008
Avaya Aura® Media Server running on Virtualized Environment	7.8.0.333
Avaya Aura Messaging	7.0
Avaya G450 Media Gateway	38.20.1
Avaya 96x1 IP Deskphones	6.65 (H323) 7.1.1 (SIP)
Talkphone VOIP-200 Series IP Call Stations	Firmware Version : 2.4.1.40

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to a Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Administer IP Node Names
- Administer Codecs
- Administer IP Network Region
- Administer Signaling Group
- Administer Trunk Group
- Administer Private Numbering
- Administer Outbound Routing

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

```
display system-parameters customer-options                               Page 1 of 10
                                OPTIONAL FEATURES

G3 Version: V16                Software Package: Enterprise
Location: 2                    System ID (SID): 1
Platform: 28                   Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 65000 290
                                Maximum Stations: 41000 44
                                Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 14
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 41000 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 **Maximum Administered SIP Trunks** supported by the system is sufficient.

```

display system-parameters customer-options                                     Page 2 of
10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                         USED
      Maximum Administered H.323 Trunks: 12000 16
    Maximum Concurrently Registered IP Stations: 18000 2
      Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
      Maximum Concurrently Registered IP eCons: 414 0
    Max Concur Registered Unauthenticated H.323 Stations: 100 0
      Maximum Video Capable Stations: 41000 1
      Maximum Video Capable IP Softphones: 18000 4
      Maximum Administered SIP Trunks: 24000 180
Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
    Maximum Number of DS1 Boards with Echo Cancellation: 522 0
      Maximum TN2501 VAL Boards: 128 0
      Maximum Media Gateway VAL Sources: 250 0
      Maximum TN2602 Boards with 80 VoIP Channels: 128 0
      Maximum TN2602 Boards with 320 VoIP Channels: 128 0
    Maximum Number of Expanded Meet-me Conference Ports: 300 0

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

```

5.2. Administer IP Node Names

Use the **change node-names ip** command to verify that node names have been previously defined for the IP addresses of the server running Communication Manager (**procr**) and for Session Manager (**interopASM**). These node names will be needed for defining the service provider signaling group in **Section 5.5**.

```

change node-names ip                                                         Page 1 of 2
                                IP NODE NAMES
      Name                IP Address
AMS1                      10.33.1.30
default                   0.0.0.0
interopASM             10.33.1.12
procr                 10.33.1.6

```

5.3. Administer Codecs

Use the **change ip-codec-set** command to define a list of codecs to use for calls between the local and remote sites. For the compliance test, codec G.711MU and G.729 was configured using ip-codec-set 1. To configure the codecs, enter the codecs in the **Audio Codec** column of the table in the order of preference, the Media Encryption section was configured to use for Avaya endpoints, the VoIP-200 station is not using the Media Encryption. Default values can be used for all other fields.

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

                                IP MEDIA PARAMETERS

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression  Per Pkt    Size (ms)
1: G.711MU      n           2          20
2: G.729      n           2          20
3:
4:
5:
6:
7:

Media Encryption      Encrypted SRTCP: enforce-unenc-srtcp
1: none
2:
3:
4:
5:
```

5.4. Administer IP Network Region

For the compliance test, IP network region 1 was chosen. Use the **change ip-network-region 1** command to configure region 1 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the local site. In this configuration, the domain name is **bwdev.com**. This name appears in the “From” header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field. This is optional.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to **yes**. This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.3**.
- Retain default values for all other fields.

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
Region: 1              NR Group: 1
Location: 1           Authoritative Domain: bwdev.com
Name: Loc-1          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
```

On **Page 4**, define the IP codec set to be used for traffic between various regions. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) **1**. Default values may be used for all other fields. In the case of the compliance test, only one IP network region was used, so no inter-region settings were required and therefore only codec set 1 is used.

```
change ip-network-region 1                                     Page 4 of 20
Source Region: 1      Inter Network Region Connection Management  I      M
                                                              G  A  t
dst codec direct  WAN-BW-limits  Video      Intervening  Dyn  A  G  c
rgn  set  WAN  Units  Total Norm  Prio Shr Regions  CAC  R  L  e
1    1
2    2    y  NoLimit                n    t
```

5.5. Administer Signaling Group

Use the **add signaling-group** command to create a signaling group between Communication Manager and Session Manager for use by SIP trunks. This signaling group is used for inbound and outbound calls between the Communication Manager and Session Manager. For the compliance test, signaling group 1 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- The compliance test was conducted with the **Transport Method** set to “tls”. The transport method specified here is used between Communication Manager and Session Manager. Whatever protocol is used here, it must also be used on the Session Manager entity link defined in **Section 6.5**.
- Set the **Peer Detection Enabled** field to “y”. The **Peer-Server** field will initially be set to **Others** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to **SM** once Communication Manager detects its peer as a Session Manager.
- Set the **Near-end Node Name** to “procr”. This node name maps to the IP address of the Communication Manager as defined in **Section 5.4**.
- Set the **Far-end Node Name** to “InteropASM”. This node name maps to the IP address of Session Manager as defined in **Section 5.2**.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a default well-known port value. (For TCP the well-known port value is 5061).
- Set the **Far-end Network Region** to the IP network region defined for the local site in **Section 5.4**.
- Set the **Far-end Domain** to the domain of the local site.
- Set **Direct IP-IP Audio Connections** to “y”. This field will enable media shuffling on the SIP trunk allowing Communication Manager to redirect media traffic directly between the SIP trunk and the enterprise endpoint.
- Set the **DTMF over IP** field to “rtp-payload”. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- Retain default values for all other fields.

SIGNALING GROUP

```

Group Number: 1                Group Type: sip
IMS Enabled? n                 Transport Method: tls
  Q-SIP? n
  IP Video? n                  Enforce SIPS URI for SRTP? n
Peer Detection Enabled? n 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: interopASM
Near-end Listen Port: 5061         Far-end Listen Port: 5061
                                     Far-end Network Region: 1

Far-end Domain: bwdev.com

Incoming Dialog Loopbacks: eliminate
                                     Bypass If IP Threshold Exceeded? n
                                     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

```

5.6. Administer Trunk Group

Use the “add trunk-group” command to create a trunk group for the signaling group created in **Section 5.6**. For the compliance test, trunk group 1 was configured using the parameters highlighted below.

- Set the **Group Type** field to “sip”.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Service Type** field to “tie”.
- Set **Member Assignment Method** to “auto”.
- Set the **Signaling Group** to the signaling group shown in **Section 5.5**.
- Set the **Number of Members** field to the number of trunk members in the SIP trunk group. This value determines how many simultaneous SIP calls can be supported by this trunk.
- Retain default values for all other fields.

```
add trunk-group 1                                     Page 1 of 22
                                                    TRUNK GROUP
Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: Private Trunk                         COR: 1                  TN: 1          TAC: #01
  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: 1
                                                    Number of Members: 14
```

On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. The **Numbering Format** was set to “private” and the **Numbering Format** in the route pattern was set to “lev0-pvt” (see **Section 5.8**).

```

add trunk-group 1                                     Page 3 of 22
TRUNK FEATURES
    ACA Assignment? n                               Measured: none
                                                    Maintenance Tests? y

    Suppress # Outpulsing? n Numbering Format: private
                                                    UUI Treatment: shared
                                                    Maximum Size of UUI Contents: 128
                                                    Replace Restricted Numbers? y
                                                    Replace Unavailable Numbers? y

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

    Send UCID? y
  
```

5.7. Administer Private Numbering

Private numbering defines the calling party number to be sent to the far-end. Use the **change private-numbering** command to create an entry that will be used by the trunk groups defined in **Section 5.6**. In the example shown below, all calls originating from a 4-digit extension beginning with “3” and routed across trunk group 1 are sent with a 4-digit calling number.

```

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

Ext  Ext      Trk      Private      Total
Len  Code     Grp(s)     Prefix       Len
4  3         1          4      Total Administered: 5
4                                     Maximum Entries: 540
  
```

5.8. Administer Outbound Routing

In these Application Notes, the Automatic Alternate Routing (AAR) feature is used to route outbound calls via the SIP trunk to the SIP endpoint. In the sample configuration, the dial prefix “34” is used as the Dialed String. This common configuration is illustrated below with little elaboration. Use the “change dialplan analysis” command to define a dialed string beginning with 34 of length 4 as extension (ext).

```

change dialplan analysis                               Page 1 of 12
                DIAL PLAN ANALYSIS TABLE
                Location: all                          Percent Full: 5

    Dialed  Total  Call   Dialed  Total  Call   Dialed  Total  Call
String    Length Type   String  Length Type   String  Length Type
34         4 ext
  
```

The route pattern defines which trunk group will be used for an outgoing call and performs any necessary digit manipulation. Use the “change route-pattern” command to configure the parameters for the local site route pattern in the following manner. The example below shows the values used for route pattern 1 during the compliance test.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** Enter the outbound trunk group for the SIP trunk. For the compliance test, trunk group **1** was used.
- **FRL:** Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of **0** is the least restrictive level.
- **Numbering Format:** “lev0-pvt”. All calls using this route pattern will use the private numbering table. See setting of the **Numbering Format** in the trunk group form in **Section 5.6** for full details.
- Retain default values for all other fields.

```

change route-pattern 1                                     Page 1 of 3
                Pattern Number: 1           Pattern Name: SIP-TLS-To-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: 1      0
2:
3:
4:
5:
6:

      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                lev0-pvt next
2: y y y y y n  n      rest                none
3: y y y y y n  n      rest                none
  
```

Use the “change aar analysis” command to create an entry in the AAR Digit Analysis Table for this purpose. The example below shows entries created for the local site “aar analysis 3”. The highlighted entry specifies that 4 digit dial string 3 was to use route pattern 1 to route calls to the SIP endpoint via Session Manager.

```

change aar analysis 3                                     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
3      4      4      1      lev0     n
  
```

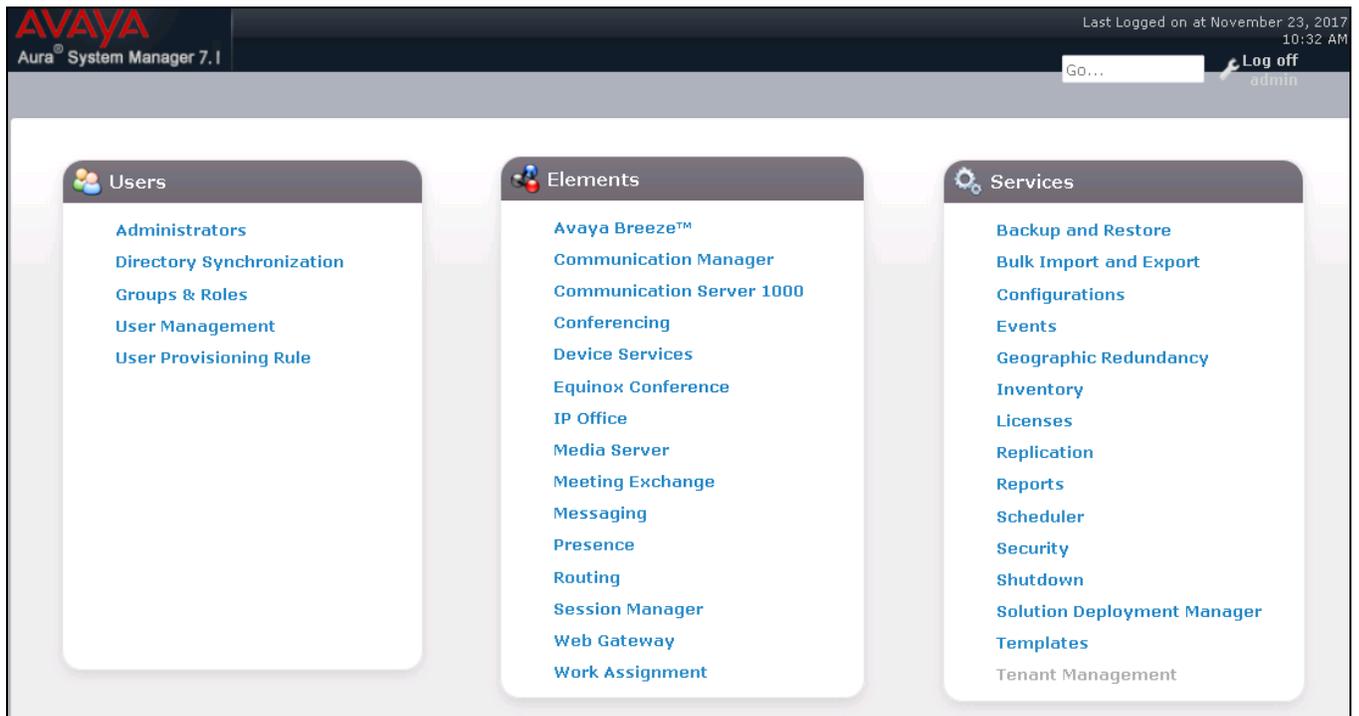
6. Configure Avaya Aura® Session Manager

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

- SIP Domain
- Location
- SIP Entities
- Entity Links
- Routing Policies
- Dial Patterns

For detail configuration details of the Session Manager refer to **Section 10**.

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in with the appropriate credentials and click on **Log on** (not shown). The following page is displayed. The links displayed below will be referenced in subsequent sections to navigate to items requiring configuration.



Clicking the **Elements** → **Routing** link, displays the **Introduction to Network Routing Policy** page. In the left-hand pane is a navigation tree containing many of the items to be configured in the following sections.

The screenshot shows the Avaya Aura System Manager 7.1 interface. The top header includes the Avaya logo, the text 'Aura System Manager 7.1', and the user's login information: 'Last Logged on at November 23, 2017 10:32 AM' and 'Log off admin'. Below the header is a breadcrumb trail: 'Home / Elements / Routing'. A navigation tree on the left lists various configuration options under the 'Routing' category, including Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Introduction to Network Routing Policy' and contains the following text:

Network Routing Policy consists of several routing applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the routing applications (that means the overall routing workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other routing applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"

- SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
- Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)

6.1. Specify SIP Domain

Create a SIP Domain for each domain for which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the domain (**bvwdev.com**) as defined in **Section 5.4**. Navigate to **Routing** → **Domains** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name** – Enter the domain name.
- **Type** – Select “sip” from the pull-down menu.
- **Notes** – Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the added domain.

The screenshot shows the Avaya Aura System Manager 7.1 interface. The left navigation pane is expanded to 'Routing', and 'Domains' is selected. The main content area is titled 'Domain Management' and shows a table with one entry. The table has columns for Name, Type, and Notes. The entry is: Name: bvwdev.com, Type: sip, Notes: SIP Domain. There are 'Commit' and 'Cancel' buttons at the top right and bottom right of the main area.

Name	Type	Notes
* bvwdev.com	sip	SIP Domain

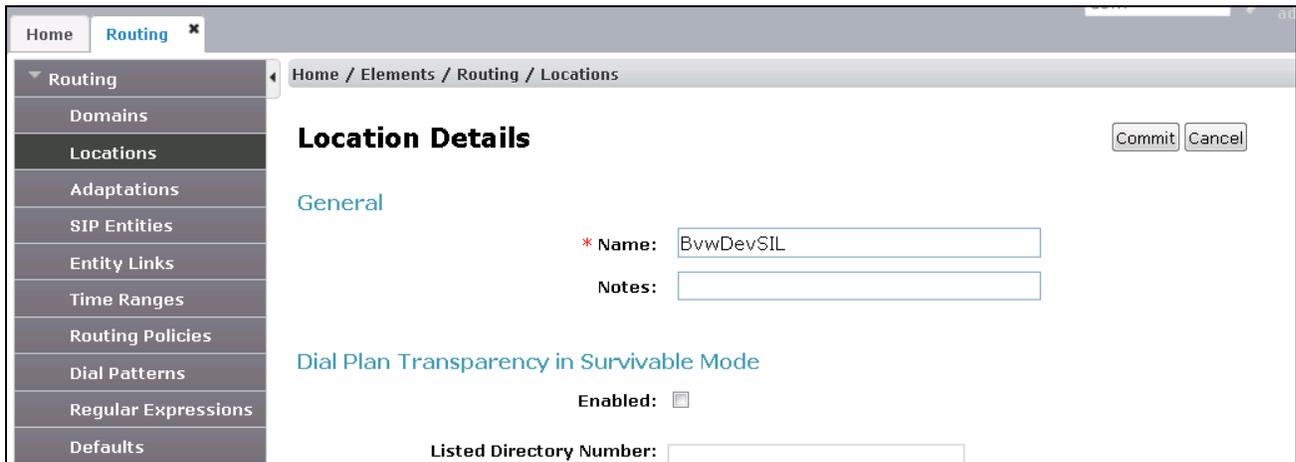
6.2. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. A single Location was defined for the enterprise even though multiple subnets were used. The screens below show the addition of the Location named **BvwDevSIL**, which includes all equipment at the enterprise including Communication Manager, Session Manager and the Dialogic SR140 PC.

To add a Location, navigate to **Routing** → **Locations** in the left-hand navigation pane (**Section 6.1**) and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Name** – Enter a descriptive name for the Location.
- **Notes** – Add a brief description (optional).

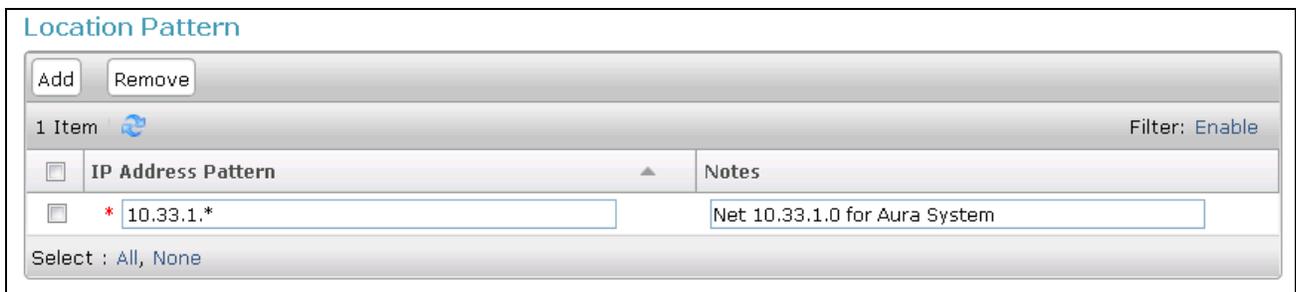


The screenshot shows a web interface for configuring a location. The left-hand navigation pane is open to 'Routing' > 'Locations'. The main content area is titled 'Location Details' and has 'Commit' and 'Cancel' buttons. Under the 'General' section, the 'Name' field is filled with 'BvwDevSIL' and the 'Notes' field is empty. Below this, the 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox that is unchecked. At the bottom, the 'Listed Directory Number' field is empty.

Scroll down to the **Location Pattern** section. Click **Add** and enter the following values.

- **IP Address Pattern** – Add all IP address patterns used to identify the location.
- **Notes** – Add a brief description (optional).

Click **Commit** to save.



The screenshot shows the 'Location Pattern' configuration page. At the top, there are 'Add' and 'Remove' buttons. Below them, it says '1 Item' and 'Filter: Enable'. A table lists the patterns:

IP Address Pattern	Notes
10.33.1.	Net 10.33.1.0 for Aura System

At the bottom, there is a 'Select : All, None' option.

6.3. Add SIP Entity

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager which includes Communication Manager. Navigate to **Routing** → **SIP Entities** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Name** – Enter a descriptive name.
- **FQDN or IP Address** – Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type** – Enter “Session Manager” for Session Manager or “CM” for Communication Manager.
- **Adaptation** – This field is only present if Type is not set to Session Manager. If applicable, select the appropriate Adaptation name. During compliance testing no adaptation rule was used.
- **Location** – Select the Location that applies to the SIP Entity being created. For the compliance test, all components were located in Location “BvwDevSIL” created in **Section 6.3**.
- **Time Zone** – Select the time zone where the server is located.

The following screen shows the addition of Session Manager. The IP address of the virtual SM-100 Security Module is entered for **FQDN or IP Address**.

The screenshot shows the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo, the text 'Aura System Manager 7.1', and a 'Last Logged on at November' status. The main navigation pane on the left lists various configuration options under the 'Routing' tab, with 'SIP Entities' selected. The main content area displays the 'SIP Entity Details' form. The form has a 'General' section with the following fields: 'Name' (ASM70A), 'FQDN or IP Address' (10.33.1.12), 'Type' (Session Manager), 'Location' (BvwDevSIL), 'Time Zone' (America/Toronto), and 'Minimum TLS Version' (Use Global Setting). There are also 'Commit' and 'Cancel' buttons at the top right of the form. A 'Monitoring' section is partially visible at the bottom of the form.

To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP Entities.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port** – Port number on which Session Manager can listen for SIP requests.
- **Protocol** – Transport protocol to be used with this port.
- **Default Domain** – The default domain associated with this port.
- **Endpoint** – Checked the checkbox to indicate the specific ports used for SIP endpoint.

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	bvwdev.com	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	5062	TLS	bvwdev.com	<input type="checkbox"/>	
<input type="checkbox"/>	5067	TLS	bvwdev.com	<input type="checkbox"/>	
<input type="checkbox"/>	5080	TCP	bvwdev.com	<input type="checkbox"/>	

Select : All, None

The following screen shows the addition of Communication Manager. In order for Session Manager to send SIP service provider traffic on a separate entity link to Communication Manager; this requires the creation of a SIP Entity for Communication Manager for use with all other SIP traffic. The **FQDN or IP Address** field is set to the IP address of Communication Manager. The **Location** field is set to **BvwDevSIL** which is the Location defined for the subnet where Communication Manager resides. See **Section 6.3**.

The screenshot displays the Avaya Aura System Manager 7.1 interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 7.1", and a "Last Logged on at November" indicator. Below the navigation bar, there are tabs for "Home" and "Routing". The left sidebar contains a tree view with the following items: Routing, Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area shows the "SIP Entity Details" page for "ACM-Trunk1-Private". The breadcrumb trail is "Home / Elements / Routing / SIP Entities". The page title is "SIP Entity Details" with a "Help ?" link. There are "Commit" and "Cancel" buttons. The "General" tab is active, showing the following fields: Name: ACM-Trunk1-Private; FQDN or IP Address: 10.33.1.6; Type: CM; Notes: Private SIP trunk for SIP phone; Adaptation: (empty dropdown); Location: BvwDevSIL; Time Zone: America/Toronto; SIP Timer B/F (in seconds): 4; Minimum TLS Version: Use Global Setting; Credential name: (empty text box); Securable: (unchecked checkbox); Call Detail Recording: both.

6.4. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. The Entity Link was created to Communication Manager, to add an Entity Link, navigate to **Routing** → **Entity Links** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name** – Enter a descriptive name.
- **SIP Entity 1** – Select the Session Manager SIP Entity.
- **Protocol** – Select the transport protocol used for this link. This must match the protocol used in the Communication Manager signaling group in Section 5.5.
- **Port** – Port number on which Session Manager will receive SIP requests from the far-end. For the Communication Manager Entity Link, this must match the one defined on the Communication Manager signaling group in Section 5.5.
- **SIP Entity 2** – Select the name of the other system. For the Communication Manager Entity Link, select the Communication Manager SIP Entity defined in Section 6.4.
- **Port** – Port number on which the other system receives SIP requests from Session Manager. For the Communication Manager Entity Link, this must match the one defined on the Communication Manager signaling group in Section 5.5.
- **Connection Policy** – Select trusted from pull-down menu.

Click **Commit** to save. The following screen illustrates the Entity Link to Communication Manager. The protocol and ports defined here must match the values used on the Communication Manager signaling group configuration in **Section 5.5**.

Home / Elements / Routing / Entity Links

Entity Links Commit Cancel [Help ?](#)

1 Item Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2
<input type="checkbox"/>	* ASM70_ACM_Trunk1_5	* ASM70A	TLS	* 5061	* ACM-Trunk1-Private

Select : All, None

6.5. Add Routing Policies

Routing Policy describes the conditions under which calls will be routed to the SIP Entities specified in **Section 6.4**. Routing Policy must be added for Communication Manager. To add a Routing Policy, navigate to **Routing** → **Routing Policies** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- Name – Enter a descriptive name.
- Notes – Add a brief description (optional).

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP Entity to which this Routing Policy applies and click **Select**. The selected SIP Entity displays on the **Routing Policy Details** page as shown below. Use default values for remaining fields. Click **Commit** to save.

The following screen shows the Routing Policy for Communication Manager.

Home / Elements / Routing / Routing Policies Help ?

Routing Policy Details

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
ACM-Trunk1-Private	10.33.1.6	CM	Private SIP trunk for SIP phone

Time of Day

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/> 0	24/7	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7						

Select : All, None

6.6. Add Dial Patterns

Dial Patterns are needed to route calls through Session Manager. For the compliance test, Dial Patterns were needed to route calls from Communication Manager to the SIP endpoint and vice versa. Dial Patterns define which Route Policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a Dial Pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane (**Section 6.1**) and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Pattern** – Enter a dial string that will be matched against the Request-URI of the call.
- **Min** – Enter a minimum length used in the match criteria.
- **Max** – Enter a maximum length used in the match criteria.
- **SIP Domain** – Enter the destination domain used in the match criteria.
- **Notes** – Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the Routing Policy from the list that will be used to route all calls that match the specified criteria. Click **Select**. Default values can be used for the remaining fields. Click **Commit** to save.

The first example shows the pattern (4 digits) that begins with “34” and has a destination domain of “bvwdev.com” from “All” location use route policy “ACM-Trunk1-Private”.

Dial Pattern Details

Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

2 Items

Filter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-		To-CM-Trunk1	0	<input type="checkbox"/>	ACM-Trunk1-Private	

6.7. Add a SIP User

A SIP user must be added for Talkphone VoIP station. Click **User Management** → **Manage Users** → **New** (not shown) and configure the following in the **Identity** tab.

- **First Name** – Enter an identifying name
- **Last Name** – Enter an identifying name
- **Login Name** – Enter the extension number followed by the domain, in this case “3409@bvwdev.com”

The screenshot shows a web browser window with the address bar displaying "Home / Users / User Management / Manage Users". The page title is "New User Profile". In the top right corner, there are buttons for "Commit & Continue", "Commit", and "Cancel", along with a "Help ?" link. Below the title bar, there are four tabs: "Identity" (marked with a red asterisk), "Communication Profile", "Membership", and "Contacts". The "Identity" tab is active and contains the following fields:

- User Provisioning Rule:** A dropdown menu.
- Identity:** A section header with a dropdown arrow.
- Last Name:** Text input field containing "SIP".
- Last Name (Latin Translation):** Text input field containing "SIP".
- First Name:** Text input field containing "3409".
- First Name (Latin Translation):** Text input field containing "3409".
- Middle Name:** Text input field.
- Description:** Text area.
- Login Name:** Text input field containing "3409@bvwdev.com".
- Email Address:** Text input field.
- User Type:** Dropdown menu with "Basic" selected.
- Password:** Text input field.
- Confirm Password:** Text input field.

Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the Talkphone VoIP station.

The screenshot shows the 'New User Profile' form with the 'Communication Profile' tab selected. The 'Communication Profile Password' and 'Confirm Password' fields are visible, both containing four dots. The 'Commit & Continue', 'Commit', and 'Cancel' buttons are at the top right.

In the **Communication Address** section select **New**; for **Type** select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number and select the appropriate Domain from the drop down list, in this case the SIP domain is “bwvdev.com”. Click **Add** when done.

The screenshot shows the 'Communication Address' section of the 'New User Profile' form. The 'Communication Profile Password' and 'Confirm Password' fields are still visible. Below them is a table with columns 'Name', 'Handle', and 'Domain'. The table is currently empty, with a 'No Records found' message. Below the table, the 'Type' dropdown is set to 'Avaya SIP', and the 'Fully Qualified Address' field contains '3409' and the 'Domain' dropdown is set to 'bwvdev.com'. The 'Add' button is visible at the bottom right.

Name	Handle	Domain
No Records found		

Select the check box for **Session Manager Profile** and configure the **Primary Session Manager, Origination Sequence, Termination Sequence** and **Home Location**, from the respective drop down lists.

Session Manager Profile ▾

SIP Registration

* Primary Session Manager

Primary	Secondary	Maximum
13	0	13

Secondary Session Manager

Survivability Server

Max. Simultaneous Devices ▾

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▾

Termination Sequence ▾

Call Routing Settings

* Home Location ▾

Conference Factory Set ▾

Call History Settings

Enable Centralized Call History?

Select the check box for **CM Endpoint Profile** and configure as follows:

- **System** – Select the relevant Communication Manager Element from the drop down list
- **Profile Type** – Select “Endpoint” from the drop down list
- **Extension** – Enter the required extension number, in this case “3409”
- **Template** – Select “9641SIP_DEFAULT_CM_7_1” from the drop down list
- **Port** – The “IP” is auto filled out by the system

CM Endpoint Profile

* System

* Profile Type

Use Existing Endpoints

* Extension

* Template

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Calculate Route Pattern

Sip Trunk

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

Continuing from above, click on **Endpoint Editor**. Click on the **Feature Options** tab, the screen shot below shows the Feature options that were used during compliance testing.

General Options (G) *	Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)	Enhanced Call Fwd (E)
Button Assignment (B)	Profile Settings (P)	Group Membership (M)		
Active Station Ringing	single			Auto Answer none
MWI Served User Type	None			Coverage After Forwarding
Per Station CPN - Send Calling Number	None			Display Language english
IP Phone Group ID				Hunt-to Station
Remote Soft Phone Emergency Calls	as-on-local			Loss Group 19
LWC Reception	spe			Survivable CDR internal
AUDIX Name	None			Time of Day Lock Table None
EC500 State	enabled			Voice Mail Number
Short/Prefixed Registration Allowed	default			
Music Source				
Features				
<input type="checkbox"/> Always Use	<input type="checkbox"/> Idle Appearance Preference			
<input type="checkbox"/> IP Audio Hairpinning	<input type="checkbox"/> IP SoftPhone			
<input type="checkbox"/> Bridged Call Alerting	<input checked="" type="checkbox"/> LWC Activation			
<input type="checkbox"/> Bridged Idle Line Preference	<input type="checkbox"/> CDR Privacy			
<input checked="" type="checkbox"/> Coverage Message Retrieval	<input checked="" type="checkbox"/> Precedence Call Waiting			
<input type="checkbox"/> Data Restriction	<input checked="" type="checkbox"/> Direct IP-IP Audio Connections			
<input checked="" type="checkbox"/> Survivable Trunk Dest	<input type="checkbox"/> H.320 Conversion			
<input type="checkbox"/> Bridged Appearance Origination Restriction	<input type="checkbox"/> IP Video			
<input checked="" type="checkbox"/> Restrict Last Appearance	<input type="checkbox"/> Per Button Ring Control			

7. Configure Talkphone VOIP-200 Series IP Call Stations

This section covers the configuration of the Talkphone VOIP-200 Series IP Call Stations. The following procedures are covered:

1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Audio Settings
5. Configure Call Parameters
6. Configure Buttons

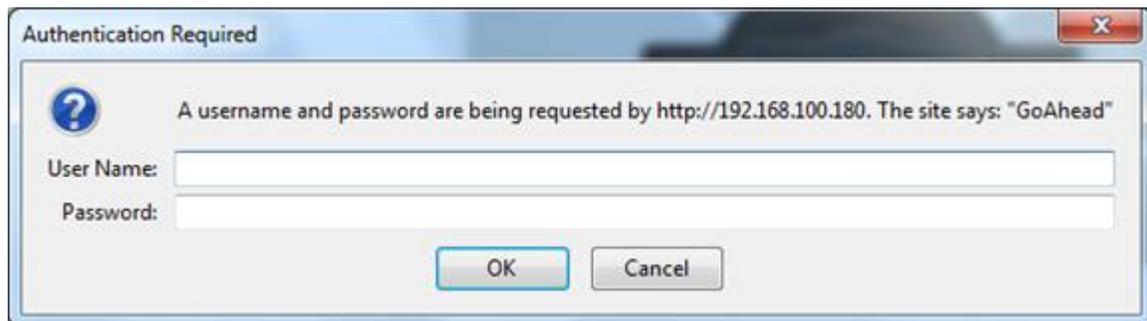
For more information on configuring other features of the Talkphone IP Call Stations, refer to [10].

7.1. Launching the Web Administration Interface

The Talkphone IP Call Stations are pre-configured with the following default values:

- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** admin@123

Ensure that the administration PC and Talkphone IP Call Station are connected to the LAN. Open a web browser and enter the IP address of the Talkphone IP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials above.



7.2. Network Configuration

To modify the IP network configuration of the Talkphone IP Call Station, navigate to the **Configuration → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Save** when done.

TALKAPHONE

Home Configuration Administration Diagnostics Network Security

IP Settings

DHCP Static IP

IP Address: 10 . 33 . 5 . 211

Subnet Mask: 255 . 255 . 255 . 0

Gateway: 10 . 33 . 5 . 1

DNS Server 1: 10 . 10 . 98 . 60

DNS Server 2: 0 . 0 . 0 . 0

Hostname: voip0000FF

Use last IP address on DHCP failure:

IGMP Version: Default

Save

7.3. SIP Configuration

Navigate to **Configuration → SIP Settings** to configure the SIP setting of the Talkphone IP Call Station. Configure the following parameters.

- **Description**
 - **Display Name** – enter a descriptive name, e.g. VoIP-200
 - **Directory Number (SIP ID)** – enter the username “3409” as configured in **Section 6.7**
 - **Primary SIP Server** – enter the SIP signaling IP address of Session Manager
 - **Username** – enter the username “3409” as configured in **Section 6.7**
 - **Password** – enter the password of the SIP user “3409”. This is the ‘Communication Profile Password’ as configured in Section 6.7.
 - **Re-registration time** – leave at the default value
 - **Outbound Proxy 1** – enter the SIP signaling IP address of Session Manager as configured in **Section 6.4**

- **Call Settings**

- **Enable auto-answer** – the VoIP-200 IP station will auto answer an incoming call if this feature enabled
- **Auto-answer Delay** – enter time in second for the auto-answer delay
- **Codec G.711 PCM u-law** – select “High Priority” in the drop-down menu

Leave other settings at default and click on **Save** button.

The screenshot shows the TALKAPHONE web interface. The top navigation bar includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar contains a menu with options: IP Settings, SIP Settings (selected), Audio Settings, Buttons, Auxiliary Output, Digital Outputs Scripts, Digital Outputs Events, Voice Messages Played to User, Voice Messages Played to Remote Side, and Time Settings.

The main content area is divided into two sections: **Registration Settings** and **Call Settings**.

Registration Settings table:

Description	Configuration
Display Name:	Voip-200
Directory Number (SIP ID):	3409
Primary SIP Server:	10.33.1.12
Secondary SIP Server:	
Tertiary SIP Server:	
Registration Method:	Parallel
Username:	3409
Password:	••••
Re-registration Time:	3600 (Range: 60-14400 seconds)
Outbound Proxy 1 (optional):	10.33.1.12 Port: 5060
Outbound Proxy 2 (optional):	Port: 5060
Outbound Proxy 3 (optional):	Port: 5060

Call Settings table:

Description	Configuration
Enable auto-answer:	<input checked="" type="checkbox"/>
Auto-answer Delay:	1 seconds (Range: 0 to 30 seconds)
Provisional Timer:	0 seconds (Range: 0 to 60 seconds) Delays call setup using input buttons
Overlap dialing:	<input type="checkbox"/>
DTMF method:	RFC 2833
Call LED off during ringing:	<input type="checkbox"/>
Hang-up on Silence Timer:	0 seconds (0 = Hang-up on Silence disabled)
Codec G.711 PCM u-law:	High Priority
Codec G.711 PCM A-law:	Low Priority
Codec G.722:	Low Priority
Codec G.729:	Low Priority

At the bottom of the settings area, there is a **Save** button.

7.4. Configure Audio Settings

Navigate to **Configuration** → **Audio Settings** to configure Speaker gain, Volume override level, Noise reduction level. All fields were left at the default values. Click **Apply** when done.

The screenshot shows the TALKAPHONE web interface. The top navigation bar includes Home, Configuration, Administration, Diagnostics, and Network Security. The left sidebar lists various settings categories, with Audio Settings selected. The main content area is titled 'Audio Settings' and contains a table with the following settings:

Description	Configuration	
Speaker gain:	5	
Volume override level:	8	Sets the volume during volume override.
Noise reduction level:	0	0 = disabled.
Microphone gain:	5	
Remote controlled volume override mode:	<input checked="" type="checkbox"/>	(DTMF * to talk, DTMF # to listen, DTMF 0 for open duplex)
Message controlled volume override mode:	<input type="checkbox"/>	(SIP Message controls audio direction)
AEC:	0	Default 0 (Restart required)
Conversation Mode:	Open Duplex	
Tone Volume:	0	Keypad and ringing tone volume
Tftp Server IP Address:		Tftp server used for downloading audio files. If empty, the station will use the configuration for TFTP remote provisioning.
Upload audio file to be played through built-in speaker for an inbound/outbound call:		Wav file to be played. The wav file must be 16 kHz, 16 bit, single channel wav file and be below 1 mb file size. It might take several minutes before the wav file is downloaded from tftp server and applied.
Play audio file through built-in speaker during an:	Outbound Call	

At the bottom of the configuration area is a 'Save' button.

7.5. Configure Buttons

Navigate to **Configuration → Buttons** to verify the appropriate settings. For the compliance test, the **Button 1** was configured as shown below. Note that the VoIP-200 only has one button.

- **Buttons (Idle)** – Button 1: enter a phone number in **Value** field to make outbound call
- **Buttons (Active Call)** – Button 1: select “Disconnect” in the dropdown menu **Function** to enable the VoIP-200 ability to disconnect the call.

Home Configuration Administration Diagnostics Network Security

- ▶ IP Settings
- ▶ SIP Settings
- ▶ Audio Settings
- ▶ Buttons
- ▶ Auxiliary Output
- ▶ Digital Outputs Scripts
- ▶ Digital Outputs Events
- ▶ Voice Messages Played to User
- ▶ Voice Messages Played to Remote Side
- ▶ Time Settings

Buttons (Idle)

	Function	Value	Option
Button 1	Call To	3306	None ▼
Button 2	Call To		None ▼
Button 3	Call To		None ▼

Buttons (Active Call)

	Function	Activated	Deactivated
Button 1	Disconnect ▼		
Button 2	Do Nothing ▼		
Button 3	Do Nothing ▼		

Numberlist Settings

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 2	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 3	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 5	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 6	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 7	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 8	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 9	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 10	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 11	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 12	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 13	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 14	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>

Call Until Answer (loops the numberlist)

Ringer Time seconds, (0=unlimited)

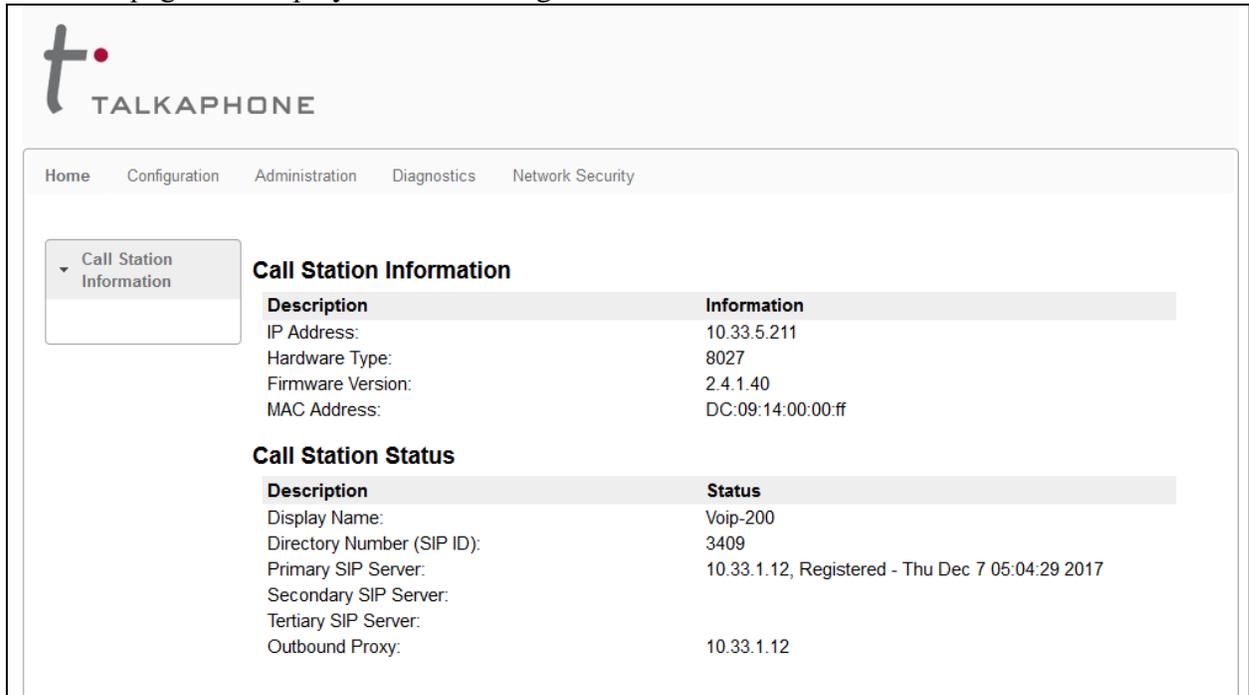
Call Conversation Timer seconds, (Range: 0 to 9999 seconds, 0 = unlimited)

Local Interdigit Timer seconds, (Range: 5 to 20 seconds)

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkaphone VOIP-200 Series IP Call Stations with Session Manager.

The 'Home' page will display the current Registration status.



The screenshot displays the Talkaphone web interface. At the top left is the logo with a stylized 't' and the word 'TALKAPHONE'. Below the logo is a navigation menu with links for Home, Configuration, Administration, Diagnostics, and Network Security. A sidebar on the left contains a dropdown menu for 'Call Station Information'. The main content area is divided into two sections: 'Call Station Information' and 'Call Station Status'. Each section contains a table with two columns: 'Description' and 'Information' or 'Status'.

Description	Information
IP Address:	10.33.5.211
Hardware Type:	8027
Firmware Version:	2.4.1.40
MAC Address:	DC:09:14:00:00:ff

Description	Status
Display Name:	Voip-200
Directory Number (SIP ID):	3409
Primary SIP Server:	10.33.1.12, Registered - Thu Dec 7 05:04:29 2017
Secondary SIP Server:	
Tertiary SIP Server:	
Outbound Proxy:	10.33.1.12

From the Talkaphone IP station makes an outbound call to a local endpoint and verifies 2-way audio and proper call termination.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Talkphone VOIP-200 Series IP Call Stations with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Talkphone IP Call Stations successfully registered with Avaya Aura® Session Manager and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

10. Additional References

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

- [1] Administering Avaya Aura® Communication Manager, Release 7.1, August 2017, Document Number 03-300509, Issue 1.
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.1, August 2017, Document Number 555-245-205, Issue 1.
- [3] Administering Avaya Aura® Session Manager, Release 7.1, Issue 1 August 2017
- [4] Administering Avaya Aura® System Manager, Release 7.1, Issue 1, August, 2017

The following Talkphone documentation may be found at www.talkphone.com.

- [5] *Talkphone VOIP-200 Series Phone Configuration and Operation Manual v3.0.2*, Rev 7/31/2012.

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