AVAYA

Avaya Solution & Interoperability Test Lab

Application Notes for Fonolo Voice Call-Backs with International Callback Support Version 3.3 with Avaya Aura® Communication Manager Release 10.1 and Avaya Aura® Session Manager Release 10.1 using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Fonolo Voice Call-Backs with International Callback Support application to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

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.

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1. Introduction

These Application Notes describe the configuration steps required for Fonolo Voice Call-Backs (Fonolo VCB) with International Call Back to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. Fonolo VCB with International Call Back provides functionality to replace hold time with a call back. During this compliance testing, Fonolo VCB with International Call Back supported two country codes +1 for US & Canada and +44 for United Kingdom. The solution communicates via SIP/RTP. The Fonolo VCB functionality was compliance tested utilizing SIP trunks to Session Manager. The configuration allowed Communication Manager to use SIP trunking for calls to and from the VCB application. The Fonolo VCB is a call center solution where instead of a caller staying in the queue when agents are all busy, caller can request to get a call back when an agent becomes available.

When a caller encounters a scenario where no agents are available in a call center environment, and Communication Manager is part of that environment, the caller is presented with options by the call center to either continue waiting in the queue or receive a call back from the call center. If the caller chose the latter, then the call center directs the caller to Fonolo VCB via a Session Manager SIP trunk where Fonolo VCB then provides a message to the caller to leave a call back number, so that Fonolo VCB can call back the caller when an agent becomes available. Once Fonolo VCB receives the confirmed call back number from the caller, Fonolo VCB uses a SIP trunk with Session Manager to call back into the call center and wait in the queue until an agent becomes available. When an agent becomes available, Fonolo VCB informs the agent that there is a call waiting and prompts if the agent would like to get connected to the caller. If the agent accepts to connect to the caller, Fonolo VCB then calls the caller via a SIP trunk to Communication Manager and connects the caller with the available agent. When Fonolo VCB makes an outbound call to the caller and agent via Session Manager, it makes two SIP INVITE requests, one to the available agent and one to the caller, and then mixes the audio within the Fonolo VCB server.

For security purposes public and lab IP addresses have been altered in this document.

2. General Test Approach and Test Results

The interoperability compliance testing focused on verifying inbound and outbound call flows between Communication Manager, Session Manager and Fonolo VCB. The feature test cases were performed manually. Calls were placed manually from users on the PSTN to a call center Vector Directory Number (VDN). During compliance testing, Call Center Elite within Communication Manager was used. An assumption was made during compliance testing in the vector script to direct callers to Fonolo VCB when no agents are available. When a caller is connected with Fonolo VCB, Fonolo VCB reads the call back number of the caller or asks the caller to input a new call back number. Fonolo VCB recognized the Dual Tone Multi Frequency (DTMF) input provided by the caller confirming the call back number. For compliance testing purposes, agents were made available after the above call between the caller and Fonolo VCB is completed. Fonolo VCB then called into the call center VDN and connected with an available agent. Fonolo VCB provided a recording, informing the agent of a call in waiting, and checked if

the agent wanted to get connected to the PSTN caller. The agent can accept the call by using DTMF input. Fonolo VCB then made the second outbound call to the PSTN caller via Communication Manager and if the PSTN caller answered the call they then get connected with the agent.

The serviceability test cases were performed manually by disconnecting and reconnecting the SIP trunk connection to Fonolo VCB.

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 these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

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

For the testing associated with these Application Notes, the interface between Avaya systems and Fonolo did not include use of any specific encryption features as requested by Fonolo.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third-party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the third-party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g., jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another and may affect the reliability or performance of the overall solution. Different network elements (e.g., session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations and evaluate if the solution is viable to be deployed in the cloud.

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The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third-party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

The Fonolo VCB application is hosted in a cloud environment by Fonolo and the VCB application was installed and synchronized with the Fonolo on-premises appliances residing in customer's side. SIP trunks were used to connect the VCB application with Communication Manager via Session Manager. The following features and functionality were covered during compliance testing:

- Establishment of SIP trunk connectivity between Fonolo VCB and Session Manager including session refresh.
- Testing of G.711MU codec.
- Incoming calls to a VDN of Communication Manager call center can be redirected to the VCB application via the SIP trunk based on vector scripting. Outgoing calls from the VCB appliance to the VDN via Session Manager when callers decide on call back. During this compliance testing, Call Center of Communication Manager was used and is not the scope of these Application Notes.
- The VCB application can make an outbound call to the caller via Session Border Controller for Enterprise who had selected the call back option and merge the call between the caller and available agent.
- DTMF transmission to ensure that options selected by the caller and agent is accepted correctly by Fonolo VCB.
- User-to-User Information (UUI) is sent from Communication Manager to the VCB application and that the same information is sent back to the agent from the VCB application.

Serviceability testing focused on verifying the ability of Fonolo VCB to recover from adverse conditions, such as the SIP trunk going down (using 'busyout' command) and reboot of Session Manager.

2.2. Test Results

All test cases were executed and passed with the following exceptions/observations:

- Fonolo VCB only supports G.711Mu codec.
- During this compliance testing, for testing of Direct IP-IP Media (shuffling), it was ensured that Avaya Deskphones were configured such that the VCB application in the on-premises application can contact them for RTP stream.

2.3. Support

Technical support on Fonolo VCB can be obtained through the following:

- **Phone:** + 1-855-366-2500 (Toll-free)
- **Web:** https://fonolo.com/contact/
- **Email:** support@fonolo.com.

3. Reference Configuration

A simulated enterprise site consisting of Communication Manager, Session Manager and System Manager were used during compliance testing. As shown in **Figure 1**, SIP trunks were used to connect two Fonolo VCB on-premises appliances with Communication Manager via Session Manager. The configuration of Fonolo VCB was done from their cloud and was synched with the Fonolo on-premises appliances via https. Avaya Session Border Controller for Enterprise was used to provide SIP connection to SIP Service Provider for external call to PSTN. A skill set queue is configured on Communication Manager with two agents belonging to this queue. The configuration allowed the enterprise site to use SIP trunking for calls to and from Fonolo VCB via the Session Manager.
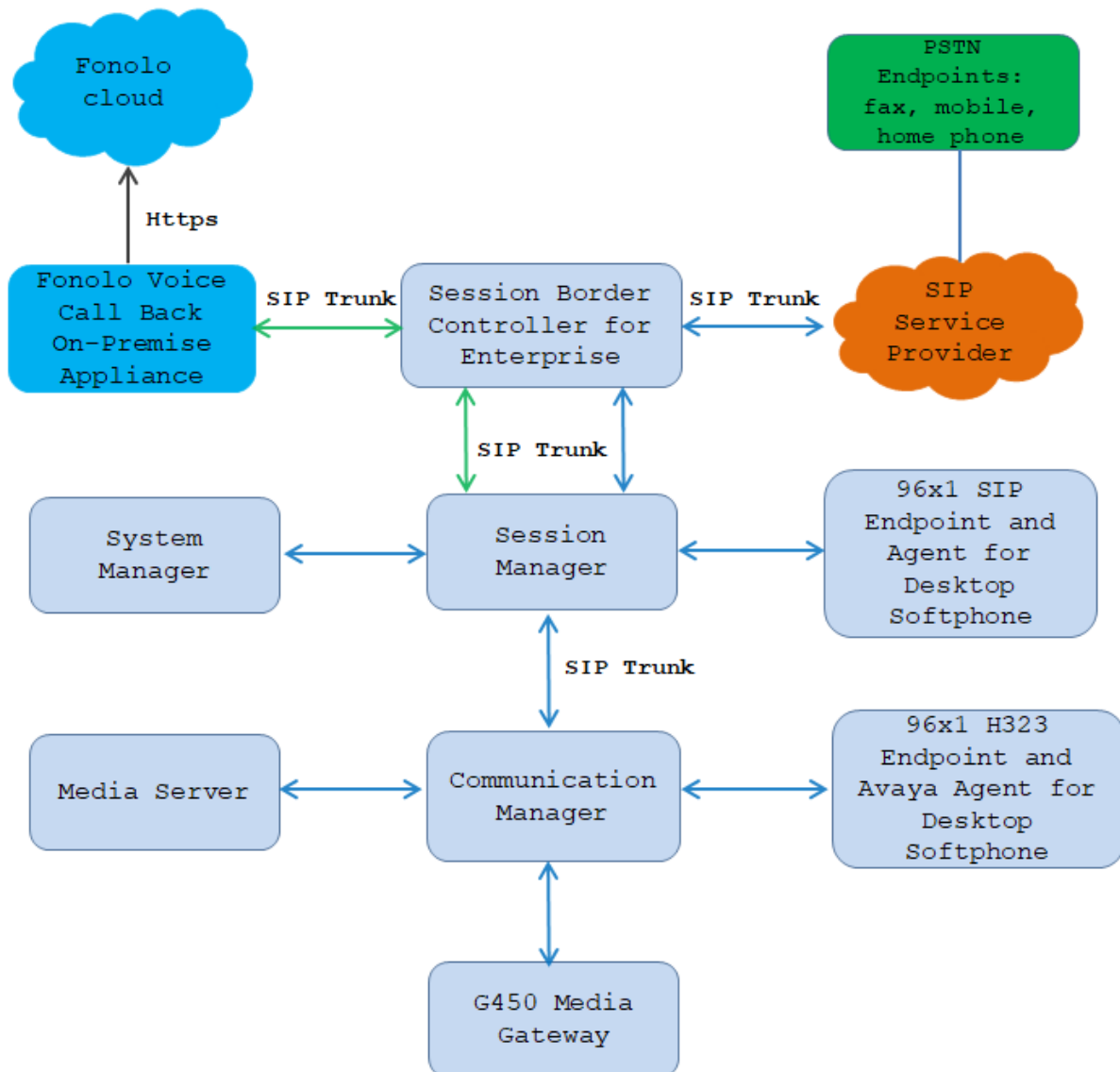
Figure 1: Reference Configuration

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 virtual server	R020x.01.0.974.0 (10.1.0.2.0.974.27607)
Avaya Aura® Media Server running on virtual server	8.0.2.218
Avaya G450 Media Gateway	42.08
Avaya Aura® System Manager running on virtual server	10.1.0.2 SP2 Software Update Revision No: 10.1.0.2.0715160 Hot Fix - 1010215160
Avaya Aura® Session Manager running on virtual server	10.1.0.2.1010215
Avaya IP Deskphones - 9641GS (H.323) - 9611G (SIP)	6.853 7.1.9.0.8
Avaya Digital 9408 Deskphone	R20
Fonolo Voice Call-Backs On-premises Appliance	Version 3.3

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The Communication Manager configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

The administration of the routing and basic connectivity between Communication Manager and Session Manager or the setting up of skill set, hunt group, vectors for a call center type environment on the Communication Manager are not the focus of these Application Notes; however, some details are provided only for informational purposes and completeness.

5.1. Verify Communication Manager License

Log into the System Access Terminal to verify that the Communication Manager license has the appropriate permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2** and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

If additional license is required, contact an authorized Avaya Sales or Reseller representative.

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

IP PORT CAPACITIES                                            USED
                  Maximum Administered H.323 Trunks: 12000      20
        Maximum Concurrently Registered IP Stations: 18000       7
          Maximum Administered Remote Office Trunks: 12000       0
Max Concurrently Registered Remote Office Stations: 18000       0
            Maximum Concurrently Registered IP eCons:  414       0
       Max Concur Reg Unauthenticated H.323 Stations:  100       0
                   Maximum Video Capable Stations:  41000       1
              Maximum Video Capable IP Softphones:  18000      12
                 Maximum Administered SIP Trunks:  40000      64
  Max Administered Ad-hoc Video Conferencing Ports: 24000       0
   Max Number of DS1 Boards with Echo Cancellation: 999     0
```

5.2. Administer IP Node Names

Use the "change node-names ip" command and add an entry for Session Manager. In this case, **SM10** and **10.33.1.42** are entered as **Name** and **IP Address**. Note the **procr** and **10.33.1.42** entry, which is the node **Name** and **IP address** for the processor board. These values will be used later to configure the SIP signaling to Session Manager in **Section 5.5**.

```
change node-names ip
                            IP NODE NAMES
    Name              IP Address
AMS1              10.33.1.44
default           0.0.0.0
SM10              10.33.1.42
lsp               10.33.1.7
procr             10.33.1.43
```

5.3. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number. Update the audio codec types in the **Audio Codec** fields as necessary. As per the observation noted in **Section** Error! Reference source not found. only configure **G.711MU**. The codec shown below was used in the compliance testing.

```
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:                    n           2         20
 3:

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

5.4. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section** Error! Reference source not found.**5**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Fonolo VCB.

```
change ip-network-region 1                                        Page   1 of  20
                                 IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1         Authoritative Domain: avayalab.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
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                    RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
           Keep-Alive Count: 5
```

5.5. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:**   Set it as "sip",
- **Transport Method:**   Set is as "tls".
- **Near-end Node Name:** Enter the "procr" interface of Communication Manager.
- **Far-end Node Name:**   Enter the node name for Session Manager.
- **Near-end Listen Port:** Enter the TLS port for the SIP trunk to Session Manager.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** Enter the existing network region to use with Session Manager.

- **Far-end Domain:** The applicable SIP domain name for the network.
- **Direct IP-IP Audio Connections?:** Set is as "y".

```
change signaling-group 1                                      Page   1 of   2
                                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                         Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                Far-end Node Name: SM10
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1


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

5.6. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** Set is as "sip".
- **Group Name:** Enter a descriptive name.
- **TAC:** Enter an available trunk access code.
- **Service Type:** Set is as "tie".
- **Signaling Group:** Enter the signaling group that has been created in **Section 5.5**.
- **Number of Members**: Enter a number of SIP trunk member, in this case 10 was used.

```
add trunk-group 1                                             Page   1 of   5
                                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: 10
```

Navigate to **Page 3** and enter "private" for **Numbering Format**.

```
add trunk-group 3                                         Page   3 of   4
TRUNK FEATURES
          ACA Assignment? n               Measured: none
                                                          Maintenance Tests? y

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

                                           Replace Restricted Numbers? y
                                          Replace Unavailable Numbers? y

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




 Show ANSWERED BY on Display? y
```

Navigate to **Page 5** and enter "y" for the **Convert 180 to 183 for Early Media?** field as shown below.

```
add trunk-group 3                                         Page   4 of   4
                        PROTOCOL VARIATIONS

                                       Mark Users as Phone? n
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n
                    Send Transferring Party Information? n
                              Network Call Redirection? y
        Build Refer-To URI of REFER From Contact For NCR? n
                                  Send Diversion Header? y
                                 Support Request History? n
                            Telephone Event Payload Type: 101


                        Convert 180 to 183 for Early Media? y
              Always Use re-INVITE for Display Updates? n
                   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
```

5.7. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Fonolo VCB. Add an entry for the trunk group defined in **Section 5.6**. In the example shown below, all calls originating from a 4-digit extension beginning with **33** and **34** and routed to trunk group **1** will result in a 4-digit calling number. The calling party number will be in the SIP "From" header.

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


Ext Ext                Trk         Private            Total
Len Code               Grp(s)      Prefix             Len
 4  33                 1                              4   Total Administered: 15
 4  34                 1                              4     Maximum Entries: 540
```

5.8. Administer Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 78xxx to Fonolo VCB. Use the "change dialplan analysis 0" command and add an entry to specify the use of digits pattern **78**, as shown below.

```
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
0            3  fac     33           4  ext     #            3  dac
1            4  ext     34           4  ext
1           11  udp     45           4  aar
78           5  udp     46           4  aar
```

5.9. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 78xxx to Fonolo VCB. Note that other routing methods may be used. Use the "change uniform-dialplan 0" command and add an entry to specify the use of AAR for routing of digits **78**xxx, as shown below.

```
change uniform-dialplan 0                                   Page   1 of   2
                     UNIFORM DIAL PLAN TABLE
                                                   Percent Full: 0


 Matching                      Insert            Node
 Pattern         Len Del       Digits        Net Conv Num
1               11  0                         ars n
35               4  0                         aar n
78               5  0                         aar n
```

5.10. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Fonolo VCB, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern Name:** Enter a descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.6**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.
- **Numbering Format**: Set to "lev0-pvt" which is private numbering plan.

```
change route-pattern 1                                       Page   1 of   4
                    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                                                        n   user
 2:                                                               n   user
 3:                                                               n   user
 4:                                                               n   user
 5:                                                               n   user
 6:                                                               n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
     0 1 2 M 4 W     Request                                Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt next
 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
```

5.11. Administer AAR Analysis

Use the "change aar analysis 78" command and add an entry to specify how to route calls to 78xxx. In the example shown below, calls with digits **78**xxx will be routed as an AAR call using route pattern "1" from **Section 010**.

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

         Dialed          Total    Route    Call  Node  ANI
         String         Min  Max  Pattern  Type  Num   Reqd
    78                    5    5    1       aar         n
```

5.12. Administer Agent Login ID

To add an **Agent LoginID**, use the command "**add agent-loginID <agent ID>**" for each agent. In the compliance test, two agent login IDs 1000 and 1001 were created.

```
add agent-loginID 1000                                     Page   1 of   2
                            AGENT LOGINID

             Login ID: 1000                                       AAS? n
                 Name: Agent 1000                               AUDIX? n
                   TN: 1
                  COR: 1
        Coverage Path:                            LWC Reception: spe
        Security Code: 1234                 LWC Log External Calls? n
            Attribute:                      AUDIX Name for Messaging:

                                           LoginID for ISDN/SIP Display? n
                                                            Password:
                                           Password (enter again):
                                                       Auto Answer: station
                                                 MIA Across Skills: system
 AUX Agent Considered Idle (MIA)? system   ACW Agent Considered Idle: system
                                           Aux Work Reason Code Type: system
                                           Logout Reason Code Type: system
                 Maximum time agent in ACW before logout (sec): system
                                           Forced Agent Logout Time:   :
    WARNING:  Agent must log in again before changes take effect
```

On **Page 2** of the **Agent LoginID** form, set the skill number (**SN**) to hunt group 1, which is the hunt group (skill) that the agents will log into.

```
add agent-loginID 1000                                     Page   2 of   2
                            AGENT LOGINID
     Direct Agent Skill:                          Service Objective? n
Call Handling Preference: skill-level          Local Call Preference? n


    SN   RL SL          SN   RL SL
 1: 1         1     16:
 2:                 17:
 3:                 18:
 4:                 19:
 5:                 20:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
```

5.13. Administer Hunt Group

This section provides the Hunt Group configuration for the call center agents. Agents will log into Hunt Group 1 configured below. Provide a descriptive name and set the **Group Extension** field to a valid extension. Enable the **ACD**, **Queue**, and **Vector** options. This hunt group will be specified in the **Agent LoginIDs** configured in **Section 5.12**.

```
add hunt-group 1                                              Page   1 of   4
                               HUNT GROUP

              Group Number: 1                                 ACD? y
                Group Name: Skill-1                           Queue? y
           Group Extension: 3320                              Vector? y
                Group Type: ucd-mia
                        TN: 1
                       COR: 1                  MM Early Answer? n
             Security Code:              Local Agent Preference? n
 ISDN/SIP Caller Display:
               Queue Limit: unlimited
 Calls Warning Threshold:       Port:
  Time Warning Threshold:       Port:
```

5.14. Administer Vector

During the compliance test, there were three vectors used for routing customer calls to the Fonolo VCB appliance. It depended on where the customer was calling from, there were two regions simulated in the testing, from US & Canada (+1) and from the United Kingdom (+44).

Use the command "change vector n" where "n" is the vector number from 1-8000. In this vector, it checks the calling number in the From header to know where the customer is calling from, if there is no calling number or anonymous, the call is routed to IVR number **4800** for more option. If the call is from UK, it routes to vector **20**. If the call is from US and Canada, it routes to vector **12**.

```
change vector 1                                               Page   1 of   6
                               CALL VECTOR

    Number: 1                     Name: Fonolo International Call
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n        Lock? n
     Basic? y    EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y    LAI? y   G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y    3.0 Enhanced? y
01 wait-time    0   secs hearing ringback
02 goto step    9             if ani          =      0
03 goto vector  20  @step 1  if ani          =      44+
04 goto vector  12  @step 1  if ani          =      1+
05 queue-to     skill 1    pri m
06 wait-time    20  secs hearing 1100     then music
07 announcement 1101
08 goto step    5             if unconditionally
09 route-to     number 4800                    cov n if unconditionally
10 busy
```

Below is vector 20 for the ANI calling from the UK with the leading digit 44. With the UK customer, the voice call-back number is set to 78001.

```
change vector 20                                            Page   1 of   6
                             CALL VECTOR

    Number: 20                  Name: Route-VCB+44
Multimedia? n     Attendant Vectoring? n    Meet-me Conf? n        Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    10   secs hearing 1100     then silence
02 goto step    10              if staffed-agents   in skill 1        =  0
03 goto step    5               if expected-wait    for skill 1   pri m  >= 0
04 queue-to     skill 1    pri m
05 collect      1    digits after announcement 1104     for none
06 set          A     = none   CATR  0123456789
07 route-to     number 78001                    cov n if digit        =  1
08 goto step    4               if available-agents in skill 1        >  0
09 stop
10 disconnect   after announcement 1106
```

Below is vector 12 for the ANI calling from US and Canada with the leading digit 1. With the US and Canada customer, the voice call-back number is set to 78000.

```
change vector 12                                            Page   1 of   6
                             CALL VECTOR

    Number: 20                  Name: Route-VCB+44
Multimedia? n     Attendant Vectoring? n    Meet-me Conf? n        Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    10   secs hearing 1100     then silence
02 goto step    10              if staffed-agents   in skill 1        =  0
03 goto step    5               if expected-wait    for skill 1   pri m  >= 0
04 queue-to     skill 1    pri m
05 collect      1    digits after announcement 1104     for none
06 set          A     = none   CATR  0123456789
07 route-to     number 78000                    cov n if digit        =  1
08 goto step    4               if available-agents in skill 1        >  0
09 stop
10 disconnect   after announcement 1106
```

5.15. Administer VDN

Use the "add vdn n" command to add a VDN number. In the **Destination** field, enter **Vector Number** 1 as configured in **Section 5.14** above and keep other fields at their default values.

```
add vdn 3340                                                   Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 3340
                                  Name*: Contact Center 1
                            Destination: Vector Number        1
                    Attendant Vectoring? n
                    Meet-me Conferencing? n
                     Allow VDN Override? n
                                    COR: 1
                                    TN*: 1
                               Measured: both     Report Adjunct Calls as
ACD*? n
          Acceptable Service Level (sec): 20
          VDN of Origin Annc. Extension*:
                              1st Skill*:
                              2nd Skill*:
                              3rd Skill*:
```

6. Configure Avaya Aura® Session Manager

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

- Launch System Manager.
- Administer Domain.
- Administer Locations.
- Administer SIP Entities.
- Administer Routing Policies.
- Administer Dial Patterns.

6.1. Launch System Manager

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

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

User ID: [   ]

Password: [   ]

[Log On]   [Reset]

ⓘ **Supported Browsers:** Internet Explorer 11.x or Firefox 58.0, 59.0 or 60.0.

6.2. Administer Domain

In the subsequent screen (not shown), select **Elements → Routing** to display the **Administration of Session Manager Routing Policies** screen below. Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new domain
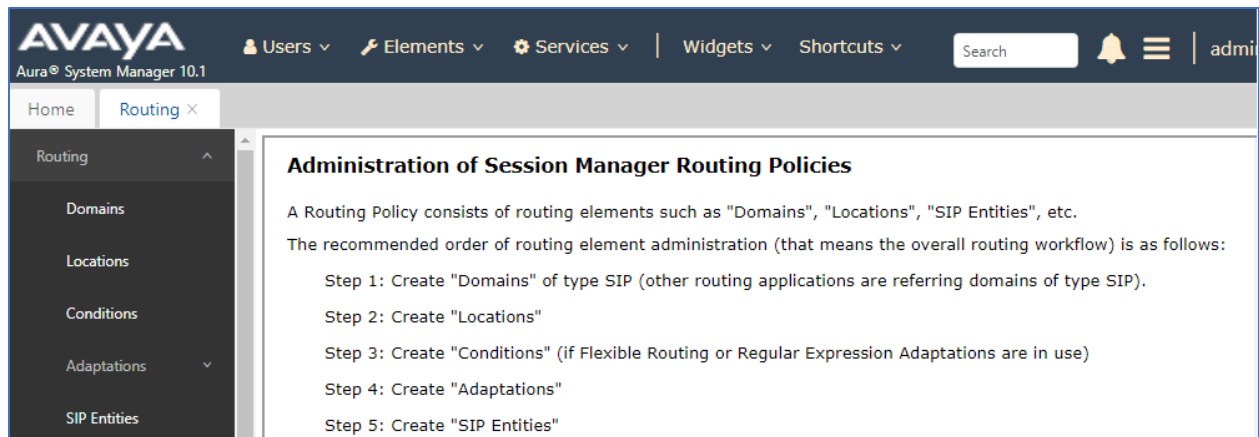
The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select "sip" from the **Type** drop down menu and provide any optional **Notes**.

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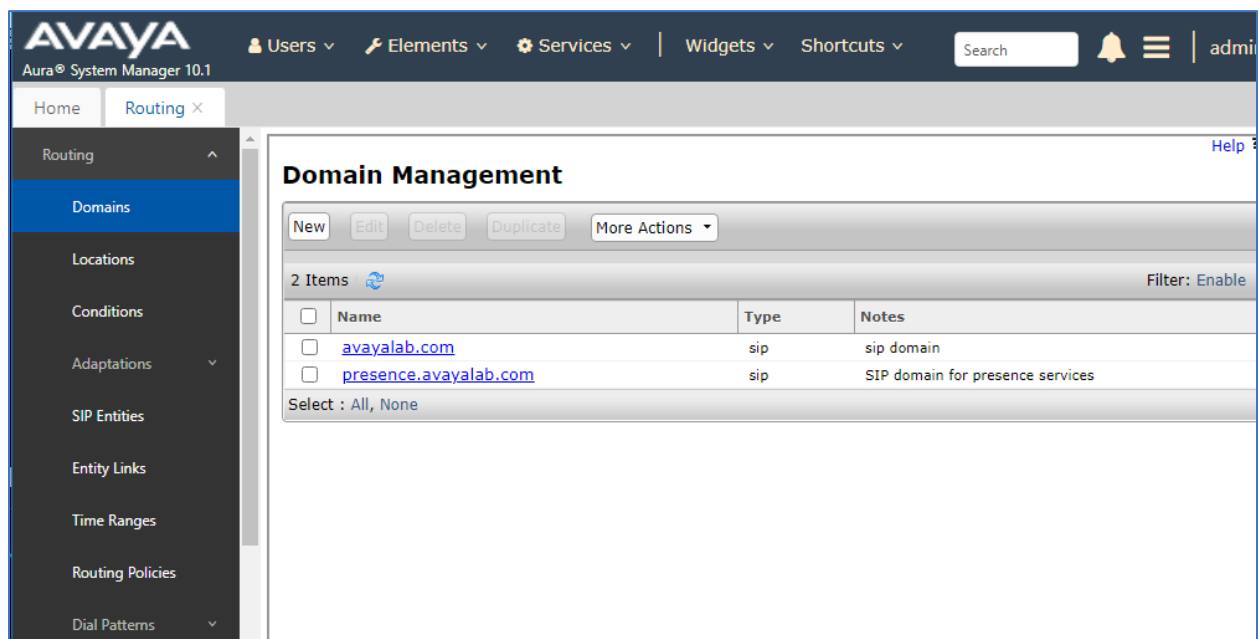
6.3. Administer Locations

Select **Routing → Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a new location for VCB.

The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

6.4. Administer Adaptation

Navigate to **Routing** → **Adaptations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Adaptation Name**: Enter an appropriate name.
- **Module Name**: Select the **DigitConversionAdapter** option.
- **Module Parameter Type**: Select **Name-Value Parameter**

Click **Add** to add the name and value parameters, as follows:

- **Name**: Enter **fromto** and set Value to **true**. Note that: the From and To headers are only modified by adaptation if the **fromto** module parameter is present and has the value of true.

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In the **Digit Conversion for Outgoing Calls from SM** section, add two entries below.
- **Matching Pattern: 1xxx, Min** and **Max: 11, Delete Digits: 0, Insert Digits: +**
- **Matching Pattern: 44xxx, Min: 12** and **Max: 14, Delete Digits: 0, Insert Digits: +**

These two adaptations are to make sure the calling number (From header) for the transfer call from Communication Manager to Fonolo VCB having the E.164 format by adding a plus sign "+" in the front of the calling number. The first entry is used for the calling numbers from US & Canada while the second one is used for the United Kingdom.

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6.5. Administer SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager, which includes Communication Manager and Fonolo VCB appliances.

6.5.1. Configure Session Manager SIP Entity

The following screen shows the previously configured Session Manager SIP Entity named **SM10**. The IP address of Session Manager's signaling interface is entered for **FQDN or IP Address 10.33.1.42**.

The ports need to be defined in Session Manager for other endpoints to connect to, scroll down to the **Listen Ports** section of the **SIP Entity Details** screen. Note that this section is only present for the **Session Manager** SIP Entity.

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

- **Port**:   Port number on which Session Manager listens for SIP requests.
- **Protocol**:   Transport protocol to be used with this port.
- **Default Domain**:   The default domain associated with this port. For the compliance test, this was the enterprise SIP Domain.

The compliance test used port **5060** for **UDP** and **5061** for **TLS** for connecting to the Fonolo appliance and Communication Manager.

Listen Ports

	Listen Ports	Protocol	Default Domain	Endpoint	Notes
☐	5060	TCP	bvwdev.com	☑	
☐	5060	UDP	bvwdev.com	☑	
☐	5061	TLS	bvwdev.com	☑	
☐	5062	TLS	bvwdev.com	☐	
☐	5067	TLS	bvwdev.com	☐	
☐	5080	TCP	bvwdev.com	☐	

Add   Remove   6 Items   Filter: Enable   Select : All, None

6.5.2. SIP Entity for Fonolo Voice Call-Backs

Select **Routing** ➔ **SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for VCB.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:**   **Enter** a descriptive name.
- **FQDN or IP Address:**   The IP address of Fonolo VCB appliance
- **Type:**   Set is as "Other".
- **Notes:**   Enter desired notes.
- **Location:**   Select the VCB location name from **Section 0**.
- **Time Zone:**   Select the applicable time zone.
- **SIP Link Monitoring:**   Select "Link Monitoring Enabled" (not shown).

Scroll down to the **Adaptations** sub-section, select **Add** button to add an entry of adaptation. Select the adaptation name "**Adding E164 Sign**" as configured in **Section 6.4** in the **Name** column.

In the **Monitoring** sub-section, leave the default option "**Use Session Manager Configuration**" in the **SIP Link Monitoring** and **CRLF Keep Alive Monitoring**.
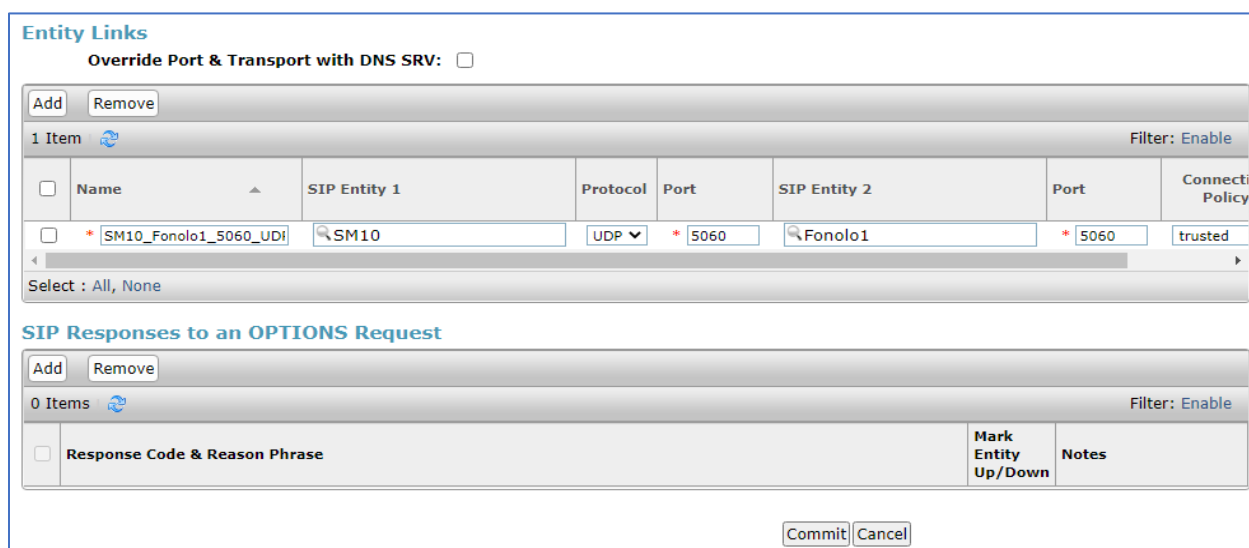
Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "**SM10**".
- **Protocol:** Set it as "UDP".
- **Port:** Set it as "5060".
- **SIP Entity 2:** The VCB entity name from this section.
- **Port:** Set it as "5060".
- **Connection Policy:** **Select** "trusted".

Note that only **UDP** protocol was tested.

Repeat the procedure above to add another SIP entity for VCB, during the compliance testing two Fonolo appliances were used for outgoing and incoming calls between Communication Manager and Fonolo appliances.

The screen below shows the adaptation "Adding E164 Sign" added.

The screen below shows the entity link for the Fonolo VCB appliance server 2.

6.5.3. SIP Entity for Communication Manager

Select **Routing** ➔ **SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that the screen below shows the previous configured SIP Entity of Communication Manager it is shown here for reference and display purpose.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:**   **Enter** a descriptive name.
- **FQDN or IP Address:**   The IP address of the processor interface.
- **Type:**   Select "CM".
- **Notes:**   Any desired notes.
- **Location:**   Select the applicable location for Communication Manager.
- **Time Zone:**   Select the applicable time zone.

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Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:**   A descriptive name.
- **SIP Entity 1:**   The Session Manager entity name, in this case "SM10".
- **Protocol:**   The signaling group transport TLS method.
- **Port:**   The signaling group listen port 5061.
- **SIP Entity 2:**   The Communication Manager entity name from this section.
- **Port:**   The signaling group listen port 5061 number.
- **Connection Policy:**   Select "trusted".

Entity Links

Override Port & Transport with DNS SRV: ☐

| Add | Remove |

1 Item 🔁   Filter: Enable

☐	Name ▲	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connec Poli
☐	* SM10_CM10_5061_TLS_I	🔍 SM10	TLS ∨	* 5061	🔍 CM10	* 5061	trusted

Select : All, None

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6.6. Administer Routing Policies

Add two new routing policies, one for VCB and one for the new SIP trunks with Communication Manager.

6.6.1. Routing Policy for Fonolo VCB

Select **Routing → Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for VCB.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Fonolo SIP entity name from **Section 6.4.2**. In the **Time of Day** sub-section, enter "0" for **Ranking**. Ranking option is only configured for the two outgoing routing policies of VCB so that calls can be load balanced. The screen below shows the result of the selection.

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6.6.2. Routing Policy for Communication Manager

Select **Routing → Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.5.33**. The screen below shows the result of the selection.

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6.7. Administer Dial Patterns

Add a new dial pattern for Fonolo VCB and Communication Manager.

6.7.1. Dial Pattern for Fonolo VCB

Select **Routing → Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach the Fonolo appliance. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "78"
- **Min:** The minimum number of digits to match
- **Max:** The maximum number of digits to match
- **SIP Domain:** The signaling group domain name from **Section 6.2**

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching the Fonolo VCB. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations "All". The VCB routing policy from **Section 6.6.1** was selected as shown below. Note that two routing policies are selected since during this compliance testing, two outgoing routing policies were configured for calls made from Communication Manager to VCB.

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6.7.2. Dial Pattern for Communication Manager

Select **Routing** → **Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Communication Manager. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "33"
- **Min:** The minimum number of digits to match
- **Max:** The maximum number of digits to match
- **SIP Domain:** The signaling group domain name from **Section 6.2**

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for call originations from all VCB endpoints in all locations. The Communication Manager routing policy from **Section 6.6.2** was selected as shown below.

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7. Configure Fonolo Voice Call-Backs

This section provides a "snapshot" of Fonolo VCB configuration used during compliance testing. Fonolo VCB is typically configured for customers by Fonolo. The screenshots and partial configuration shown below, supplied by Fonolo, are provided for reference only. These represent only an example of the configuration GUI of VCB, available through the Fonolo Customer Portal at https://portal.fonolo.com/. Other configurations are possible. Contact Fonolo for details on how to configure VCB. The configuration operations described in this section can be summarized as follows:

- Add a New SIP Trunk Group.
- Adding the Agent Call-Back Endpoint.
- Adding a New Call-Back Profile.

7.1. Add a New SIP Trunk Group

Navigate to **Telco → SIP Trunks** and click the **Add New SIP Trunk Group** at the top of the page. Define a new label to identify this SIP trunk group. During compliance testing **Avaya SM Aura 10.1 Testing** was used as the label. Then select **Add New SIP Trunk** (not shown).
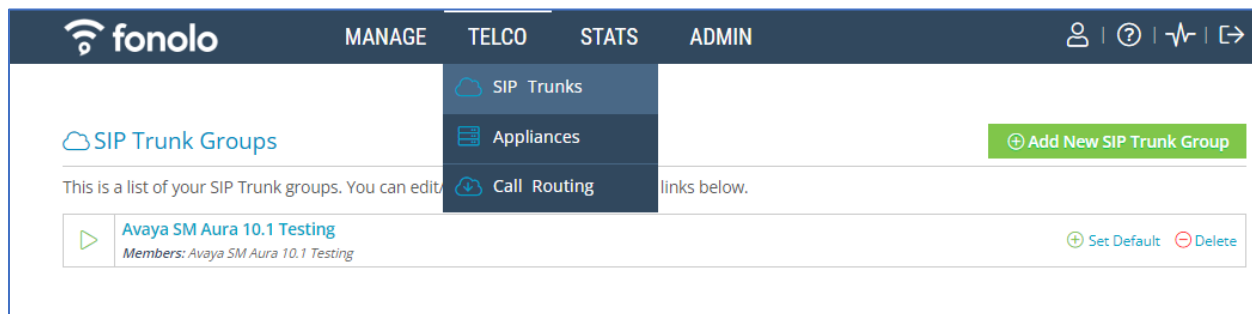
Under the **Members** tab in this new SIP trunk group, click the **Add New Member** button (not shown), and the **Add New SIP Trunk** dialog will appear as shown below.

Under **Add New SIP Trunk**:

- **SIP URL**: The IP address of Session Manager formatted as a fully qualified URL, defining the protocol and SIP port.
- **DTMF Mode**: The mode to use for sending DTMF tones. Default is RFC 2833.
- **Identity Header**: Whether to include an identity header (either Remote-Party-ID or P-Asserted-Identity). Default is none.
- **Codec Support**: The list of audio codecs to use. Default is μ-law.
- **Priority**: A numeric value that can be used to determine failover or load balance groups when more than one SIP trunk group member is defined. Members with lower priority values are used first; members with an equal priority values are load balanced.
- **Keepalive**: This instructs the Fonolo platform to perform regular keep-alive using SIP OPTIONS requests, based on the number of seconds defined. Default is disabled.

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- **Session Timers**: If Fonolo should enable SIP Session Timers (RFC 4028). Default is disabled.
- **NAT Support**: If the SIP trunk group member specified is located behind a NAT (Network Address Translation) device. Fonolo can compensate for the un-reachable RTP data specified in the SDP body of the INVITE request, using symmetric RTP.

Add the IP address of Session Manager, formatted as a fully qualified URL, defining the protocol and SIP port, then click the **Save Trunk** button. During compliance testing, the protocol **UDP** and port **5060** is used for the SIP service with Session Manager, and the default values for the remaining SIP trunk group member settings.

Update SIP Trunk

SIP Trunk SID:	TM9ce0431733d25cbcef46285d7f0392bb
SIP Label:	Avaya SM Aura 10.1 Testing — *Only visible through this interface.*
SIP URL:	udp://10.33.1.42:5060 — *SIP URL to connect to this SIP trunk member.*

*SIP URLs should use IP addresses or hostnames, and include a protocol (udp, tcp, or tls), and a port value. For example: **udp://10.10.10.10:5060***

DTMF Mode:	RFC 2833 (Recommended) ⌄ — *How we send/receive DTMF tones with this host.*
Identity Header:	None ⌄ — *If we should add an additional SIP identity header.*
From Domain:	☑ avayalab.com — *Use a custom From domain on this SIP Trunk member.*
Codec Support:	☑ μ-law ☐ a-law
Priority:	10 — *Lower priority trunks are used first. Equal priority trunks are load balanced.*
Keepalive:	☑ *Enable a keepalive timer on this host. (SIP OPTIONS)*
Session Timers:	☑ *Enable SIP Session Timers (RFC 4028) on this host.*
NAT Support:	☐ *This host is behind a NAT device.*
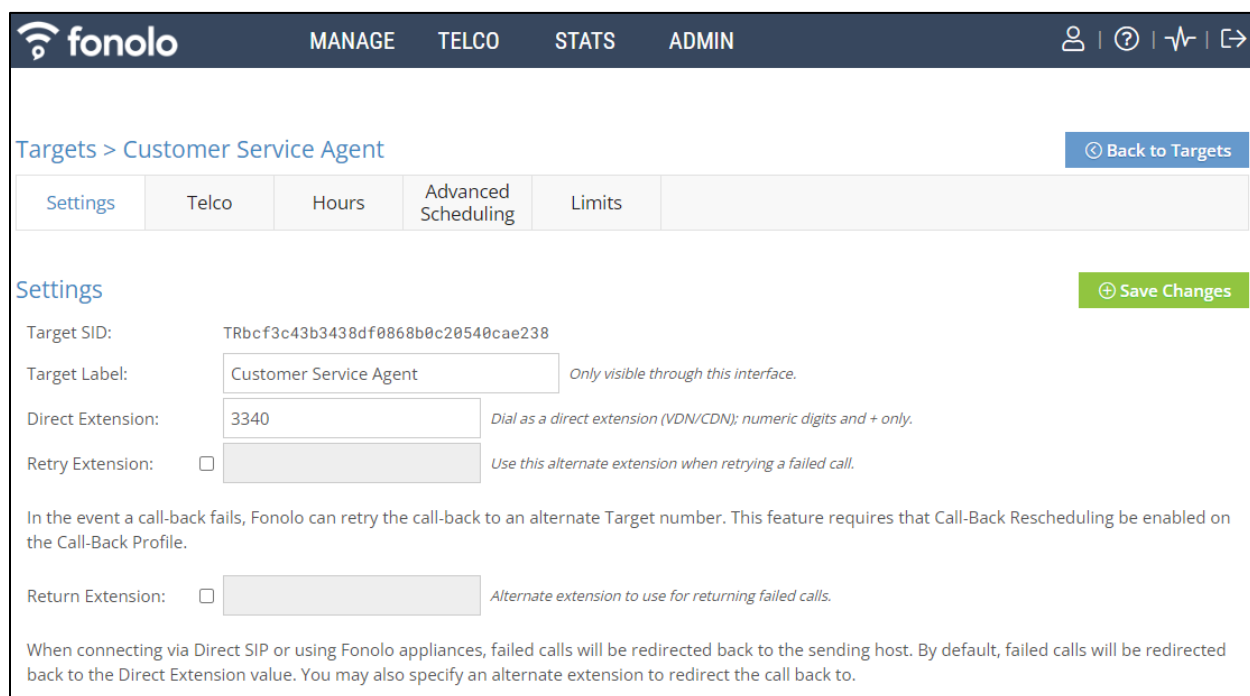
Save Trunk | Cancel

7.2. Adding the Voice Call-Back Endpoint

Navigate to **Manage → Targets** and click the **Add New Target** button. Define a new label to identify this new Target. During compliance testing, **Customer Service Agent** was used as the **Target Label**. Select the **Dial as SIP Extension** option (shown below) for **Dial Method** and enter the VDN to reach the pertinent skillset via Session Manager in the **Extension** field.
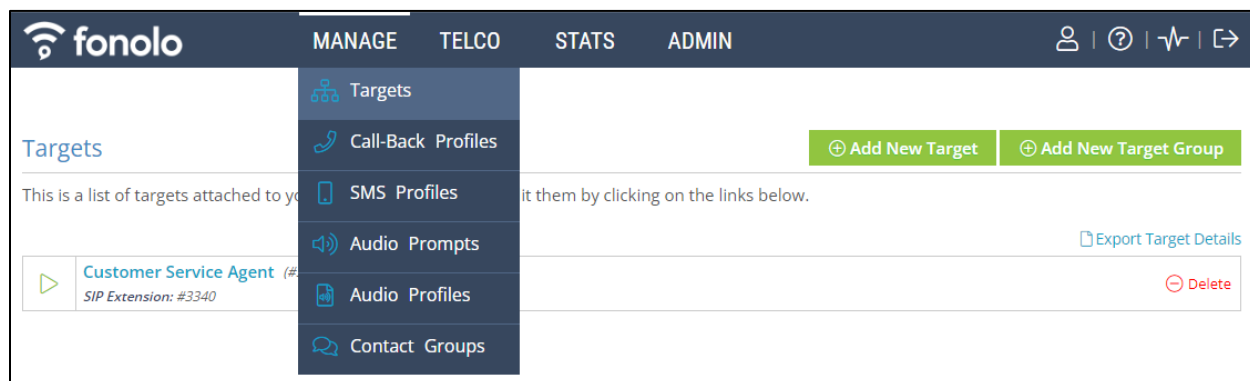
During compliance testing, VDN **3340** was pre-configured on Communication Manager which was accessible via Session Manager. Then click on the **Add New Target** button to save this Target.

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From the **Telco Settings** section of the newly added Target, select the SIP trunk to use for this Target, from the **Direct SIP** drop down menu shown below. Select the **Avaya Session Manger** SIP trunk, added in **Section** Error! Reference source not found., and then click the **Save Changes** button.

7.3. Adding a New Call-Back Profile

Navigate to **Manage → Call-Back Profiles** and click on the **Add New Profile** button (not shown), and configure the new profile:

- **Profile Label:**   A label to identify this new profile
- **Geo Whitelist:**   A geographic whitelist to use for this new profile
- **Channel:**   Select "In-Call Rescue"
- **Language:**   Select the appropriate language for this skill set queue
- **Client CID Number:**   The Caller-ID number the customer will see
- **Client CID Name:**   The Caller-ID name the customer will see
- **Agent CID Number:**   The Caller-ID number the agent will see
- **Agent CID Name:**   The Caller-ID name the agent will see

Note that there were two Call-Back profiles created for two different regions: US & Canada and the United Kingdom.

Click the **Add New Call-Back Profile** button to add this new profile.

From the **Call Options** section of the new **Call-Back Profile**, select the Target added in **Section Error! Reference source not found.** (from the drop-down menu highlighted below), and click the **Add Option** link to add the VDN value to the section on the left, as shown below, then click the **Save Changes** (not shown) button.

This associates the Target VDN with this new **Call-Back Profile**. Multiple call options can be associated with a single **Call-Back Profile**, one for each skill call-backs are being offered on. Note that there were two Call Options created for two different regions: US & Canada and the United Kingdom.

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The table below shows the Call Settings for Call Option US & Canada and Call Option UK. The Fonolo Extension for each call option is different, for (+1) US & Canada it was assigned to 78000 and (+44) UK to 78001.

Call Option Settings for US & Canada	Call Option Settings for the UK
Option Settings Option SID: C03fd55f3ff255f0f04343f26d8aa60ab9 Option Label: Customer Service Agent Fonolo Extension: 78000 The Fonolo Extension value is auto-allocated for each call-back option, and should not normally be changed. Please contact Fonolo Support for more details. ☑ Override the default country code for this option? Dialing Area: (+1) United States, Canada, & Island N ⌄ ☐ Override the client caller ID values for this option? Client CID Number: Client CID Name: ☐ Override the agent Caller ID values for this option? Agent CID Number: {{$client_number}} Agent CID Name: Fonolo [Save Option] [Cancel]	**Option Settings** Option SID: C0d22bd153c405dbc38cae78291d47ff7d Option Label: Customer Service Agent Fonolo Extension: 78001 The Fonolo Extension value is auto-allocated for each call-back option, and should not normally be changed. Please contact Fonolo Support for more details. ☑ Override the default country code for this option? Dialing Area: (+44) Great Britain (United Kingdom) ⌄ ☐ Override the client caller ID values for this option? Client CID Number: Client CID Name: ☐ Override the agent Caller ID values for this option? Agent CID Number: {{$client_number}} Agent CID Name: Fonolo [Save Option] [Cancel]

From the **Telco Settings** section of the new **Call-Back Profile**, select the **Avaya Session Manager** SIP trunk group created in **Section** Error! Reference source not found. as the **Direct SIP** value under both the **Client Call-Back Method** and the **In-Call Rescue Call Transfers** section, as shown below, then click the **Save Changes** button.

Navigate to **Manage → Call-Back Profiles** and click on the **Call Options** link on the newly created **Call-Back Profile** (not shown). The **ICR Settings** dialog will appear (shown below) and include the inbound extensions to use for VDN. These are the extensions to transfer calls to, on the VCB system, when a call opts-in for a call-back. During compliance testing, the extension **78000** and **78001** were configured on the Fonolo system.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager and Fonolo VCB.

8.1. Verify Avaya Aura® Communication Manager

On Communication Manager, verify the status of the SIP signaling group by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section Error! Reference source not found.5**. Verify that the signaling group is **in-service** as indicated in the **Group State** field shown below.

```
status signaling-group 1
                        STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

      Group State: in-service
```

Verify the status of the local SIP trunk group by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.6**. Verify that all trunks are in the **in-service/idle** state as shown below.

```
status trunk 1

                          TRUNK GROUP STATUS

Member     Port    Service State         Mtce Connected Ports
                                         Busy
0001/0001 T000001 in-service/idle        no
0001/0002 T000002 in-service/idle        no
0001/0003 T000003 in-service/idle        no
0001/0004 T000004 in-service/idle        no
0001/0005 T000005 in-service/idle        no
0001/0006 T000006 in-service/idle        no
0001/0007 T000007 in-service/idle        no
0001/0008 T000008 in-service/idle        no
```

The following tests were also performed to verify proper configuration of Fonolo VCB with Communication Manager.

- PSTN caller can select the call back option and get redirected to VCB via Communication Manager/Session Manager.
- PSTN caller can hear the VCB menu and make the required choices.
- VCB can recognize the choices made by the PSTN user.
- VCB can call the VDN and wait for an available agent.
- VCB can call out to the PSTN caller and connect them to an available agent.

8.2. Verify Avaya Aura® Session Manager

Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** and select the Communication Manager SIP Entity. Verify the **Link Status** is **Up**.

Repeat the same procedure selecting each Fonolo VCB SIP Entity and verify the **Link Status** is **Up**.

8.3. Verify Fonolo Voice Call-Back

In the Fonolo customer portal, verify the link status of the SIP trunk group to Session Manager, by navigating to **Telco → Appliances** and select the appliance group (not shown) and then select the **Member** tab. All appliances should be synched successfully.

Additional information is available through the **Stats → Graphs** section or **Stats → Call Details** section.
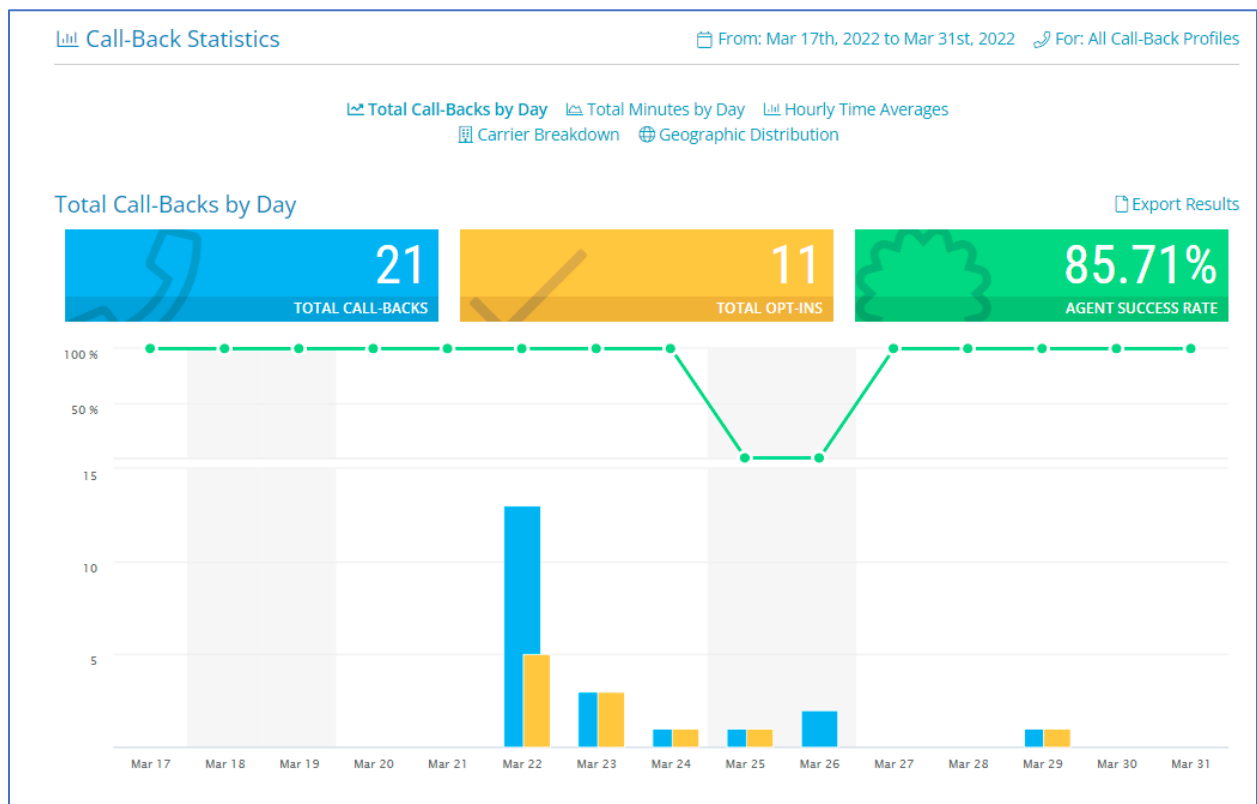
fonolo   MANAGE   TELCO   STATS   ADMIN

⊞ Call-Back Details

From: Oct 8th, 2022 to Nov 4th, 2022   For: All Call-Back Profiles

This section lets you view and export detailed Call-Back details.

Export Results

Date	Time	Option	Client	Deep Dial	Hold Time	CC Time	Talk Time	Duration	Debug Recording	Status	Clearing	Ended By	Reso
2022-10-31	13:17:30	Customer Service Agent	441423394300	00:00	01:19	00:05	00:20	01:44	--	Success	Normal Clearing	Company	
2022-10-31	13:16:15	Customer Service Agent	441423394300	00:00	01:00	00:00	00:00	01:00	--	Failed	Agent Did Not Accept	Company	
2022-10-31	13:15:00	Customer Service Agent	441423394300	00:00	01:00	00:00	00:00	01:00	--	Failed	Agent Did Not Accept	Company	
2022-10-31	13:13:30	Customer Service Agent	441423394300	00:00	01:00	00:00	00:00	01:00	--	Failed	Agent Did Not Accept	Company	
2022-10-31	13:12:20	Customer Service Agent	441423394300	00:00	01:00	00:00	00:00	01:00	--	Failed	Agent Did Not Accept	Company	
2022-10-31	13:11:15	Customer Service Agent	441423394300	00:00	01:00	00:00	00:00	01:00	--	Failed	Agent Did Not Accept	Company	
2022-10-20	01:25:36	Customer Service Agent	441423394305	00:00	01:58	00:13	132:53	135:03	--	Success	Normal Clearing	Client	
2022-10-20	01:24:21	Customer Service Agent	441423394305	00:00	01:00	00:00	00:00	00:59	--	Failed	Agent Did Not Accept	Company	
2022-10-20	01:23:51	Customer Service Agent	441423394300	00:00	01:34	00:12	127:02	128:47	--	Success	Normal Clearing	Client	
2022-10-19	12:55:51	Customer Service Agent	441423394300	00:00	01:27	00:22	00:34	02:23	--	Success	Normal Clearing	Company	
2022-10-19	12:16:10	Customer Service Agent	16139675071	00:00	01:13	00:06	30:58	32:18	--	Success	Normal Clearing	Company	
2022-10-18	11:05:36	Customer Service Agent	441423394305	00:00	01:09	00:02	67:16	68:27	--	Success	Normal Clearing	Client	
2022-10-18	11:04:50	Customer Service Agent	16139675071	00:00	01:03	00:08	68:46	69:56	--	Success	Normal Clearing	Company	
2022-10-17	11:30:24	Customer Service Agent	16139675071	00:00	00:13	00:10	14:51	15:13	--	Success	Normal Clearing	Company	
2022-10-17	11:08:50	Customer Service Agent	441423394305	00:00	01:10	00:05	18:50	20:06	--	Success	Normal Clearing	Client	

9. Conclusion

These Application Notes describe the configuration steps required for Fonolo Voice Call-Backs with International Callback Support to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in **Section** Error! Reference source not found..

10. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya product documentation, including the following, is available at http://support.avaya.com

1. *Deploying Avaya Aura® Session Manager and Branch Session Manager in Virtualized Environment*, Release 10.1, Issue 3, September 2022
2. *Administering Avaya Aura® Session Manager*, Release 10.1, Issue 3, August 2022
3. *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 10.1, Issue 4, August 2022
4. *Administering Avaya Aura® System Manager for Release 10.1*, Release 10.1, Issue 5, September 2022

5. *Deploying Avaya Aura® Communication Manager in Virtualized Environment*, Release 10.1, Issue 4, September 2022
6. *Administering Avaya Aura® Communication Manager*, Release 10.1, September 2022
7. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 10.1, September 2022
8. *Administering Avaya G430 Branch Gateway*, Release 10.1, Issue 3, September 2022
9. *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0, October 2022

Fonolo provides their documentation upon delivery of their products/services.