Avaya Solution & Interoperability Test Lab

Application Notes for FCS Voice (SIP) v3.1 with Avaya Aura® Communication Manager R8.0 and Avaya Aura® Session Manager R8.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the FCS Voice (SIP) v3.1 to interoperate with Avaya Aura® Communication Manager R8.0 and Avaya Aura® Session Manager R8.0. FCS Voice interface between Avaya Aura® Communication Manager with Session Manager and FCS Gateway, a Property Management System. It supports both SIP and analog technology. In this Compliance Testing, only the SIP interface is used.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 procedures for configuring FCS Voice (SIP) v3.1 to interoperate with Avaya Aura® Communication Manager R8.0 and Avaya Aura® Session Manager R8.0. FCS Voice (SIP) v3.1 hereafter referred to as FCS Voice in short connects to both Avaya Aura® Communication Manager with Avaya Aura® Session Manager and FCS Gateway, a Property Management System (PMS).

FCS Voice supports standard Hospitality feature requests to/from a PMS (guest room check-in/check-out/move, Automatic Wake-Up (AWU), Message Waiting Lamp (MWL) control and Housekeeping/Room Status changes and Minibar usage as well as auto attendant function. Access to these services is via SIP Trunk link from Avaya Aura® Communication Manager through Avaya Aura® Session Manager and/or the FCS Voice web interface.

2. General Test Approach and Test Results

Feature functionality testing was performed manually. Inbound and outbound voice calls were made to the guest telephones from local extensions and simulated PSTN. A simulated PMS application instead of FCS Gateway, was also used to make room check in /check out /move requests and MWL lamp On/Off for voice and text messages.

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 the FCS Voice did not include use of any specific encryption features as requested by FCS.

2.1. Interoperability Compliance Testing

Interoperability compliance testing focused on the ability of FCS Voice to work with Communication Manager and Session Manager. FCS Voice features and capabilities that were verified included the following:

- Leave and retrieve voice messages for both guest and admin phones.
- Message Waiting Light for both guest and admin phones.
- Set up and receive Automatic Wake Up Call for guest phones.
- Redirect failed Wake Up Call to Operator.
- Receive specific numbers for service calls like express message leave and retrieve, and setting wake up calls.
- Operator transfer for wakeup call failure notification as well as when caller elects not to leave a message and presses 0 instead.
- Changing Mailbox PIN and recording personal greeting.
- Using G.711Mu Law, G.711A Law and G.729 codec.

2.2. Test Results

All executed test cases were completed successfully.

2.3. Support

For technical support on FCS Voice, contact FCS Computer Systems at:

Email: helpdesk.fcs@planet1world.com
Tel: +632-672-7860

3. Reference Configuration

The configuration used in performing compliance testing of FCS Voice is shown in **Figure 1**. It shows a network consisting primarily of a pair of Communication Manager in duplex mode with an Avaya G430 Media Gateway, a System Manager and Session Manager, a FCS Voice server installed including PMS simulator. Each guest room has a pair of phones which are either analog or Avaya digital phone and an IP Telephone. Additional utility phones are setup to function as Operator, Admin and Message Desk. The SIP trunk link from FCS Voice is connected via the Session Manager which acts as proxy to Communication Manager.

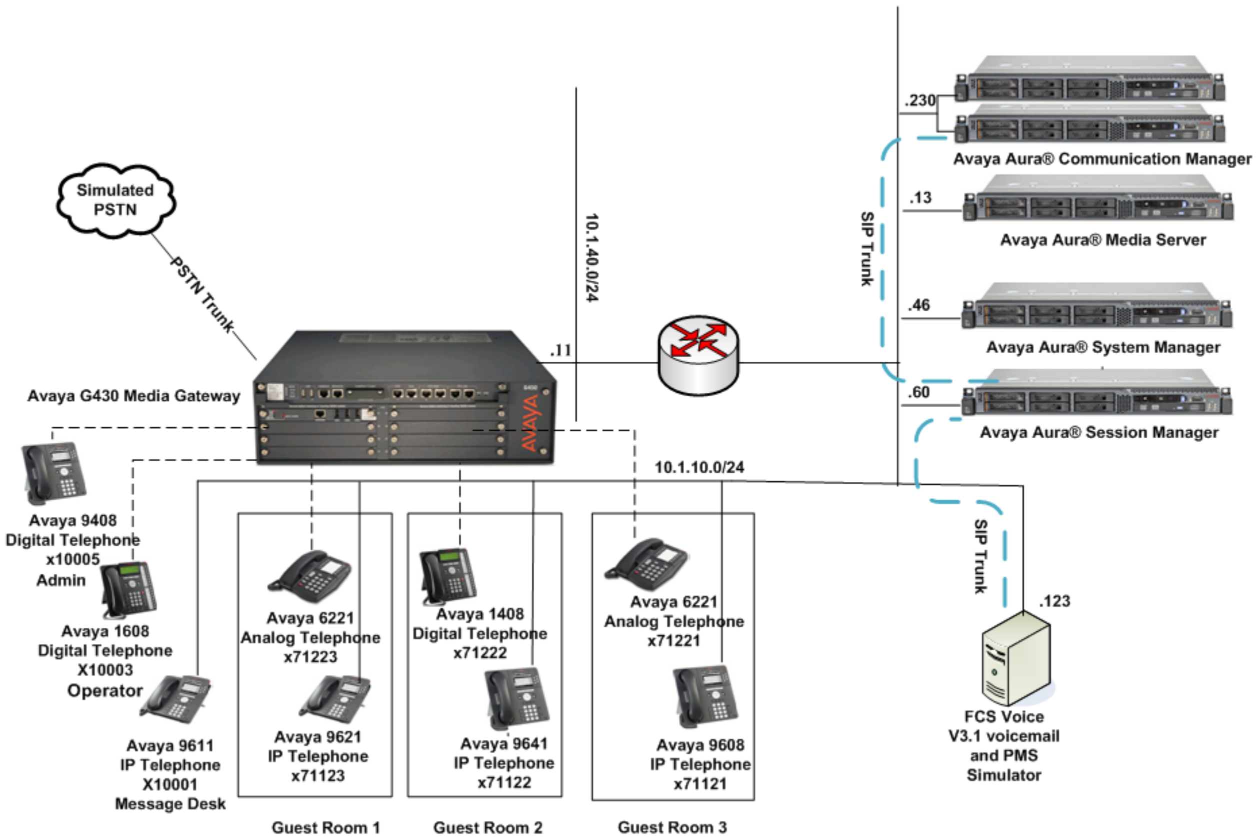


Figure 1: Sample Test Configuration

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	R018x.00.0.822.0 - 24796
Avaya G430 Media Gateway • MGP	 40.10.0
Avaya Aura® Media Server	8.0.0.150
Avaya Aura® System Manager	8.0.0.0 Build No. - 8.0.0.0.931077 Software Update Revision No: 8.0.0.0.098174
Avaya Aura® Session Manager	8.0.0.0.800035
Avaya 96x1 IP H323 Telephone	6.6604
Avaya 16xx IP H323 Telephone	1.3100
Avaya 6221 Analog Telephone	-
Avaya 14xx Digital Telephone	R4 SP10
Avaya 94xx Digital Telephone	2.0 SP4 (R15)
FCS Voice on Windows Server 2012 R2	3.1

Note: The Avaya Aura® servers including FCS Voice server used in the test configuration and shown on the table were deployed on a virtualized environment. These servers ran as virtual machines over VMware® platforms.

5. Configure Avaya Aura® Communication Manager

This section details the steps required to configure Avaya Communication Manager to interoperate with FCS Voice. These Application Notes assume the Avaya Media Gateway (including modules) has already been administered. Please refer to **[1]-[2]** for additional details. Since PMS simulator was used for this compliance testing, administration for PMS is not documented here.

The commands listed in this section were issued at the System Access Terminal (SAT) screen. For all steps where data are modified, submit the completed administration form for the changes to take effect.

5.1. License

Ensure that license is provided for the SIP Trunking to FCS Voice other than the hospitality features are turned on as below:

- **Maximum Administered SIP Trunks** : Ensure sufficient number of SIP Trunks allocated
- **IP Trunks?** Must be enabled for IP Trunks
- **ISDN-PRI?** Must be enabled for IP Trunks

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

IP PORT CAPACITIES                                                USED
                     Maximum Administered H.323 Trunks: 12000 90
           Maximum Concurrently Registered IP Stations: 18000 6
             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 0
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 30000 28
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 688   0







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

```
display system-parameters customer-options                      Page   5 of  12
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                             IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                       ISDN Feature Plus? n
                  Enhanced EC500? y       ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                         ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                ISDN-PRI? y
              ESS Administration? y              Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? y                 Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? y                 Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? y              Multimedia IP SIP Trunking? y
                       IP Trunks? y

           IP Attendant Consoles? y
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Define Session Manager as an IP Node Name

Enter **change node-names ip** and add an entry for Session Manager using an appropriately descriptive value for the **Name** (in this case, *sm1*) and the corresponding **IP Address** (in this example, *10.1.10.60*).

```
change node-names ip s                                          Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
s8500-clan1         10.1.10.21
s8500-clan2         10.1.10.22
s8500-medpro1       10.1.10.31
s8500-medpro2       10.1.10.32
s8500-val1          10.1.10.36
site6               10.1.60.18
sm1                 10.1.10.60
sm2                 10.1.10.42









( 8  of 32   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Add Client Room Properties to a Class of Service

Enter **change cos**, and for the Class of Service to be assigned to guest telephones, set the **Client Room** field to *y* (as shown below for Class of Service 5).

```
change cos-group 5                                              Page   1 of   2
CLASS OF SERVICE         COS Group: 5   COS Name: Guest

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls             n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                   n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling               n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions            n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  n  n  n  n  y  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Automatic Exclusion            n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

5.4. Set Guest Room Calling Party Restrictions in a Class of Restriction (COR)

Enter **change cor *n***, where ***n*** is the number of the Class of Restriction to be assigned to guest telephones (in this example, **COR** *5* is used).

```
change cor 5                                                    Page   1 of  23
                              CLASS OF RESTRICTION

               COR Number: 5
          COR Description: Guest Room

                      FRL: 0                                APLT? y
  Can Be Service Observed? n            Calling Party Restriction: all-toll
Can Be A Service Observer? n             Called Party Restriction: none
        Time of Day Chart: 1        Forced Entry of Account Codes? n
         Priority Queuing? n                 Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? n
     Restricted Call List? n                  Can Change Coverage? n
   Unrestricted Call List:
            Access to MCT? y             Fully Restricted Service? n
Group II Category For MFC: 7             Hear VDN of Origin Annc.? n
         Send ANI for MFE? n              Add/Remove Agent Skills? n
            MF ANI Prefix:               Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? n
                                    Can Use Directed Call Pickup? n
                                    Group Controlled Restriction: inactive
```

5.5. SIP Trunk to Session Manager

This section details the setup of the SIP trunk for calls to Session Manager where FCS Voice are routed to. It includes the following:

- Create IP Network Region and Codec
- Create Signalling-Group
- Add Sip Trunk-Group
- Create Uniform Dialplan
- Private Numbering
- Routing of Voice Mail calls

5.5.1. Create IP Network Region and Codec

Enter **change ip-codec-set 6** and setup the appropriate codec acceptable by the FCS Voice Server. In this example, *G.711Mu* and *G.711Alaw* audio codec is administered for IP Network Region 6 assigned in this compliance test for FCS Voice Server. Leave the rest as default. Codec *G.729B* was also tested.

```
change ip-codec-set 6                                          Page   1 of   2

                          IP Codec Set

    Codec Set: 6

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

Enter **change ip-network-region 6** to check that the **Codec Set** is set to *6* above.

```
change ip-network-region 6                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 6       NR Group: 6
Location: 1       Authoritative Domain: sglab.com
    Name: To Session Manager 6  Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 6              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
```

5.5.2. Create Signaling-Group

Enter **add sig n,** where **n** is the number of the signaling group created (in this example, signaling-group **7**). Enter the following parameters:

- **Group Type :** Enter *sip*
- **Transport Method :** Enter *tls*
- **Peer Detection Enabled :** Enter *y*
- **Peer Server :** This will be automatically detected as *SM* after submission of the form.
- **Near-end Node Name:** Enter *procr*
- **Near-end Listen Port:** Enter *5061*
- **Far-end Node Name:** Enter *sm1*
- **Far-end Listen Port:** Enter *5061*
- **Far-end Network Region:** Enter *6*
- **Far-end Domain:** In this case *sglab.com*

```
add signaling-group 7                                          Page   1 of   2
                                SIGNALING GROUP

Group Number: 7              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? y          Priority Video? y        Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM                        Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: sm1
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 6

Far-end Domain: sglab.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
```

5.5.3. Add SIP Trunk-Group

Enter **add trunk n,** where **n** is the number of the trunk group created (in this example, trunk-group **7**). Enter the following parameter:

- **Group Name :** Enter appropriate name
- **Group Type :** Enter **sip**
- **Service Type :** Enter **tie**
- **Signaling Group:** Enter **7**
- **Number of Members:** Enter appropriate value
- **Numbering Format:** Enter *private*
- **Support Request History:** Enter *y*
- **Telephone Event Payload Type:** Enter *101*

```
add trunk-group 7                                               Page   1 of  21
                                TRUNK GROUP

Group Number: 7                   Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk to SM1           COR: 1        TN: 1        TAC: #07
   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: 7
                                                     Number of Members: 14
```

```
add trunk-group 7                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



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

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n

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




 Show ANSWERED BY on Display? y
```

```
add trunk-group 7                                            Page   4 of  21
                              PROTOCOL VARIATIONS

                           Mark Users as Phone? n
                 Prepend '+' to Calling Number? n
           Send Transferring Party Information? n
                      Network Call Redirection? n
                         Send Diversion Header? n
                       Support Request History? y
                  Telephone Event Payload Type: 101


                       Convert 180 to 183 for Early Media? n
                 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
```

5.5.4. Create Uniform Dialplan

The Voice Mail Pilot Number 70000 is setup on FCS Voice in **Section 7.3**. This needs to be created with uniform dialing to dial the number without aar access code. Enter **change uniform-dialplan 7** to create the uniform dial plan for 70000. At the **Matching Pattern** *70000*, enter the **Len** as *5* and the **Net** as *aar*.

```
change uniform-dialplan 7                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert               Node
  Pattern       Len Del      Digits      Net Conv Num
 70000           5   0                   aar  n
```

5.5.5. Private Numbering

Enter **change private-numbering 5** to set guest rooms number as private numbering format. In this test, digit *7* is the starting digit of the guest room numbers. This is required in order for FCS Voice to obtain the history info of the guest rooms.

```
change private-numbering 5                                      Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private           Total
Len Code             Grp(s)     Prefix            Len
 5  1                6                            5     Total Administered: 5
 5  1                7                            5        Maximum Entries: 540
 5  2                10                           5
 6  4                7                            6
 5  7                7                            5
```

5.5.6. Routing of Voice Mail calls

Enter **change aar analysis x** for routing of the Voice Mail Pilot Number 70000 calls to FCS Voice server. Enter the values for **Dialed String** for *70000* as below. **Call Type** is set as *lev0* to indicate private numbering for calling number to Voice Mail with the **Route Pattern** *6* to be set in the next command.

```
change aar analysis 4                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 0

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    5                     4    4     6        lev0         n
    6                     5    5     10       aar          n
    68731233              8    8     30       pubu         n
    68731267              8    8     30       pubu         n
    70000                 5    5     6        lev0         n
```

Enter **change route-pattern 6** and enter the trunk group number under the column **Grp No** as *7* created in **Section 5.5.3. Numbering Format** is set as *lev0-pvt* to set private numbering for calling number to FCS Voice Server.

```
change route-pattern 6                                          Page   1 of   3
                    Pattern Number: 6   Pattern Name: non-IMS to SM6
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 7    0                    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  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 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
```

5.6. Create Service Numbers for Voice

The following service numbers are created for FCS Voice which is used to invoke the services:

S/No	Service Numbers	Description
1.	70001	Voice Mail message retrieval
2.	70002	Express Leave Voice Mail message
3.	70003	Set Wake Up call

Note: *The above is just an example – Voice services are configurable via the WebUI.*

The corresponding settings on FCS Voice are detailed in **Section 7.4**.

Enter **add vdn 70001** and set the appropriate **Name.** Enter **Destination** to Vector Number **71**.

```
add vdn 70001                                                Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 70001
                                 Name*: Voicemail Service 1
                           Destination: Vector Number        71
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none     Report Adjunct Calls as ACD*? n


        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:



* Follows VDN Override Rules
```

Enter **change vector 71** and set the following with the **route-to number** *70000*. This is repeated for VDN 70002 to 70003. Note the route-to number will be the same for all the VDNs listed below.

```
change vector 71                                             Page   1 of   6
                                  CALL VECTOR

    Number: 71              Name: Voicemail Service 1
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 route-to     number 70000                    cov n if unconditionally
```

The following list the VDNs that are created and correspondingly points to Vector Number 71, 72,and 73 respectively.

```
list vdn 70000 count 10

                          VECTOR DIRECTORY NUMBERS

                                                                    Evnt
                                      VDN         Vec          Orig    Noti
Name (22 characters)   Ext/Skills     Ovr COR TN  PRT Num  Meas Annc    Adj

Voicemail Service 1    70001            n 1   1    V  71   none

Voicemail Service 2    70002            n 1   1    V  72   none

Voicemail Service 3    70003            n 1   1    V  73   none
```

5.7. Creating Voice Mail Hunt Group

Enter **add hunt-group 70** and set the appropriate name. Enter *grp-name* for **ISND/SIP Caller Display**. On the next page, enter **Message Center** as *sip-adjunct*, enter **Voice Mail Number** as *70000*, **Voice Mail Handle** as *70000* and the **Routing Digits** as *8*.

```
add hunt-group 70                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 70                               ACD? n
              Group Name: FCS Voice                      Queue? n
         Group Extension: 70000                         Vector? n
              Group Type: ucd-mia                Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: grp-name
```

```
add hunt-group 70                                               Page   2 of  60
                                  HUNT GROUP




                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     70000                    70000                     8
```

5.8. Creating Default Coverage Path

The default coverage path is created here for Voice Mail coverage**.** Enter **change coverage path 70** and enter the **Point1** as *h70* (coverage hunt group 70 created in **Section 5.7**). Enter the appropriate **Number of Rings** so that it is longer than the time for the automatic wake-up to consider as no answer if it goes into coverage. Otherwise, repeat Wake Up call will not function. Refer to **Section 7.2** for the FCS Voice *Auto Wakeup Ringing Duration*. In this compliance test, the **Number of Rings** is set to *3*.

```
change coverage path 70                                         Page   1 of   1
                                 COVERAGE PATH

                               Coverage Path Number: 70
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
             Active?            n              n
               Busy?            y              y
       Don't Answer?            y              y        Number of Rings: 3
                All?            n              n
 DND/SAC/Goto Cover?            y              y
   Holiday Coverage?            n              n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h70           Rng:    Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

5.9. Assign Class of Service and Class of Restriction Values to Guest Telephones

For each guest telephone extension ***x***, enter **change station *x*** and enter in the **COR** and **COS** fields the values corresponding to the Class of Service and Class of Restriction administered in **Section 5.3 and 5.4**, respectively. Enter **Coverage Path 1** as *70*. In actual cases where PMS link is setup, the coverage path will be set by PMS and this is configured via the **change system hospitality** form which is not covered in this document.

On the next page, set the **MWI Served User Type** as *sip-adjunct* and turn on the **Per Station CPN –Send Calling Number** to *y*.

```
change station 71121                                            Page   1 of   4
                                     STATION

Extension: 71121                         Lock Messages? n              BCC: 0
     Type: 1608                          Security Code: 111222          TN: 1
     Port: S00195                      Coverage Path 1: 70             COR: 5
     Name: 71121                       Coverage Path 2:                COS: 5
                                       Hunt-to Station:              Tests? y
STATION OPTIONS
                Location:                  Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 71121
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
                              Short/Prefixed Registration Allowed: default
```

```
change station 71121                                            Page   2 of   4
                                     STATION
                                 FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed                        EC500 State: enabled
         Multimedia Mode: enhanced                   Audible Message Waiting? n
    MWI Served User Type: sip-adjunct             Display Client Redirection? n
                                                 Select Last Used Appearance? n
                                                   Coverage After Forwarding? s
                                                     Multimedia Early Answer? n
                                                 Direct IP-IP Audio Connections? y
  Emergency Location Ext: 71121         Always Use? n IP Audio Hairpinning? N
```

On the last page, set the **voice-mail** as *70000* for speed dial access via the MESSAGE button and the appropriate room number for **Room**.

```
change station 71121                                          Page   4 of   4
                                     STATION
 SITE DATA
       Room: Room 3                                   Headset?
       Jack:                                          Speaker? n
      Cable:                                          Mounting: d
      Floor:                                       Cord Length: 0
   Building:                                         Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                           5:
 2: call-appr                           6:
 3: call-appr                           7:
 4:                                     8:

    voice-mail 70000
```

6. Configure Avaya Aura® Session Manager

This section describes the procedures for configuring Session Manager to support the routing of calls to FCS Voice server.

These instructions assume other administration activities have already been completed such as defining SIP entities for Session Manager, defining the network connection between System Manager and Session Manager, and defining Communication Manager as a Managed Element. Please refer to **[3]-[4]** for additional details.

The following administration activities will be described:

- Define SIP Domain and Locations
- Define SIP Entity for FCS Voice Server
- Define Entity Links, which describe the SIP trunk parameters used by FCS Voice Server when routing calls between SIP Entities
- Define Routing Policies and Dial Patterns which control routing between SIP Entities

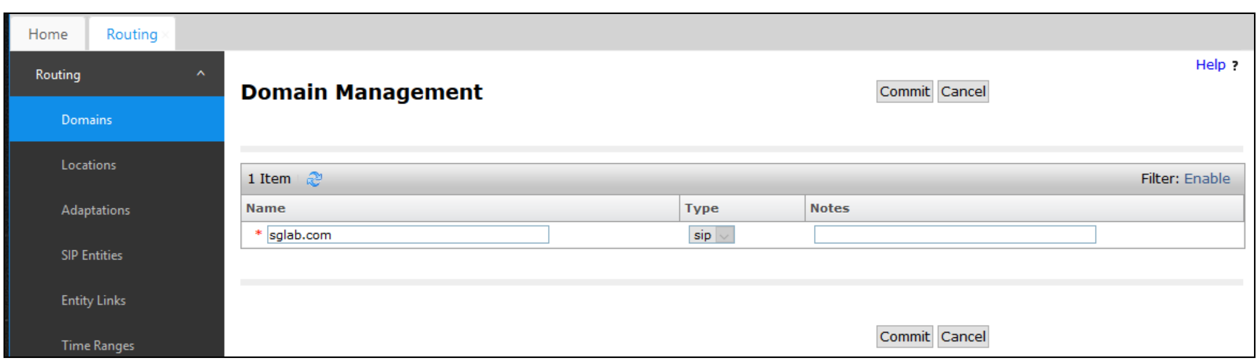
Configuration is accomplished by accessing the browser-based GUI of Avaya System Manager, using the URL "**http://<ip-address>/SMGR**", where "**<ip-address>**" is the IP address of Avaya System Manager. Log in with the appropriate credentials.

6.1. Define SIP Domains

Expand **Elements** →**Routing** and select **Domains** from the left navigation menu. Click **New** (not shown) and enter the following values and use default values for remaining fields**.**

- **Name** Enter the Authoritative Domain Name
  For the sample configuration, "**sglab.com**" was used.
- **Type** Select "**sip**" from drop-down menu.
- **Notes** Add a brief description. [Optional].

Click **Commit** to save. The screen below shows the SIP Domain defined for the sample configuration.



6.2. Define Locations

Locations are used to identify logical and/or physical locations where SIP Entities or SIP endpoints reside, for purposes of bandwidth management or location-based routing. Expand **Elements →Routing** and select **Locations** from the left navigation menu.

Click **New** (not shown)**.** In the **General** section, enter the following values and use default values for remaining fields.

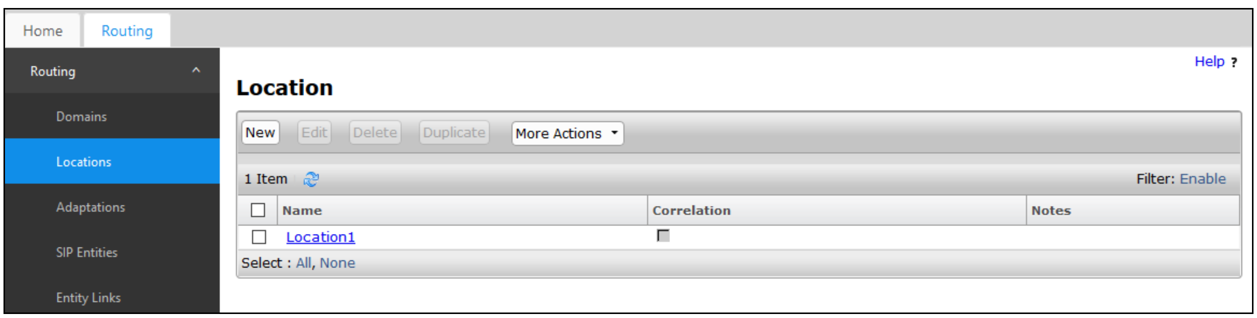
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description. [Optional].

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

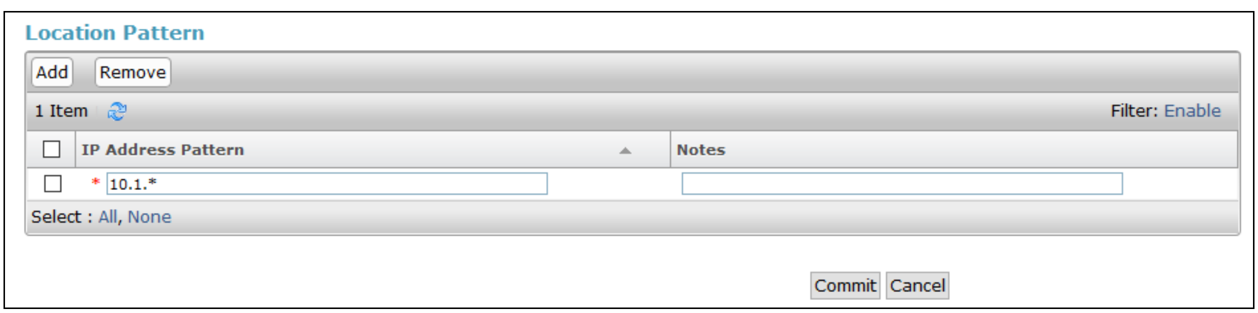
- **IP Address Pattern** Enter the logical pattern used to identify the location. For the sample configuration, "**10.1.***" was used.
- **Notes** Add a brief description. [Optional]

Click **Commit** to save.

The screen below shows a *Location1* used for SIP entities in the sample configuration.



Note: screen has been abbreviated for clarity.



6.3. Define SIP Entities

A SIP Entity must be added for FCS Voice Server. To add a SIP Entity, expand **Elements →Routing** and select **SIP Entities** from the left navigation menu.

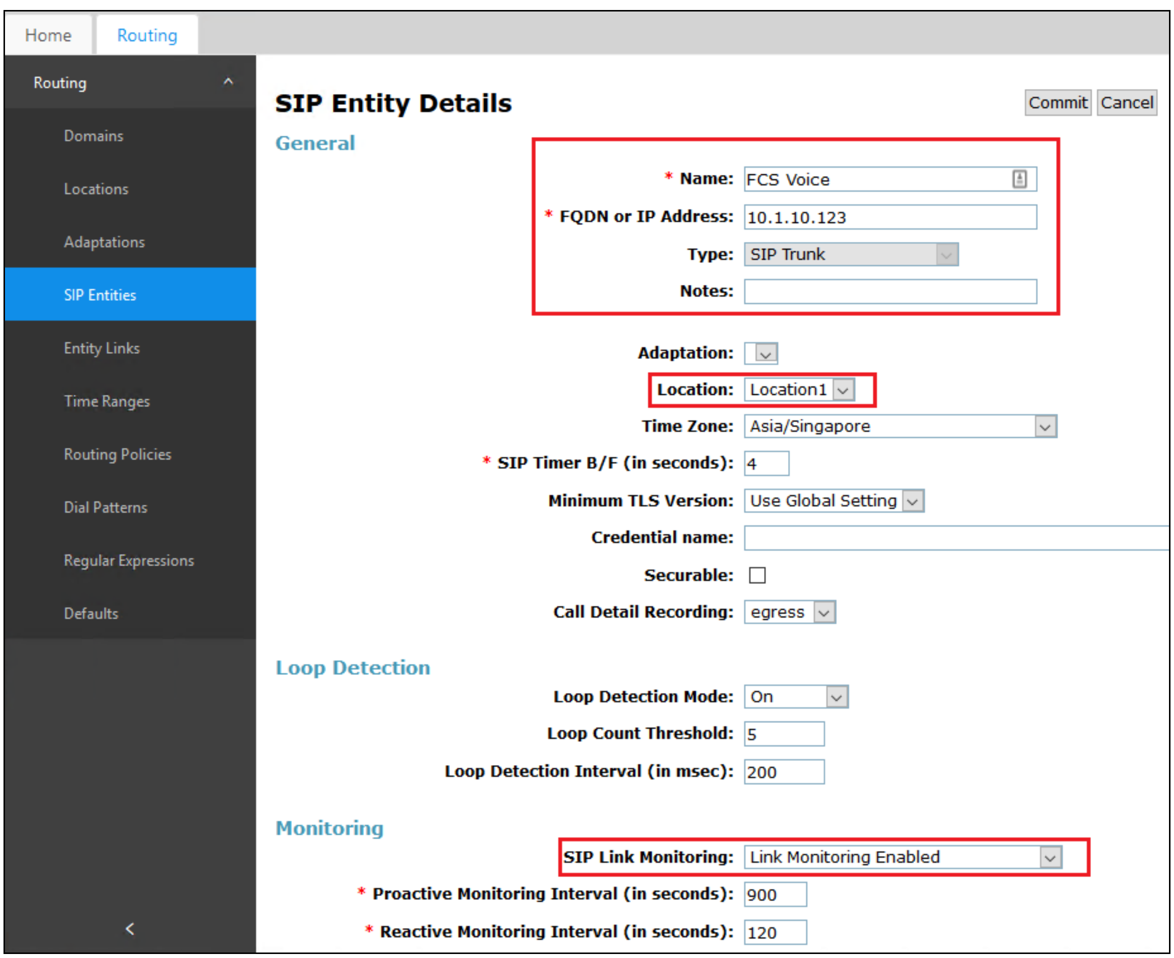
Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for new SIP Entity.
  In the sample configuration, *FCS Voice* was used.
- **FQDN or IP Address:** Enter IP address as *10.1.10.123*.
- **Type:** Select *SIP Trunk*.
- **Notes:** Enter a brief description. [Optional].
- **Location:** Select Location defined for Communication Manager in **Section 6.2**.

In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select *Link Monitoring Enabled*. This is because FCS Voice supports OPTION request for status.

Click **Commit** to save SIP Entity definition. The following screen shows the SIP Entity defined for FCS Voice.



6.4. Define Entity Links

A SIP trunk between FCS Voice Server and Session Manager is described by an Entity Link. In the sample configuration, SIP Entity Links were added between Session Manager and FCS Voice Server.

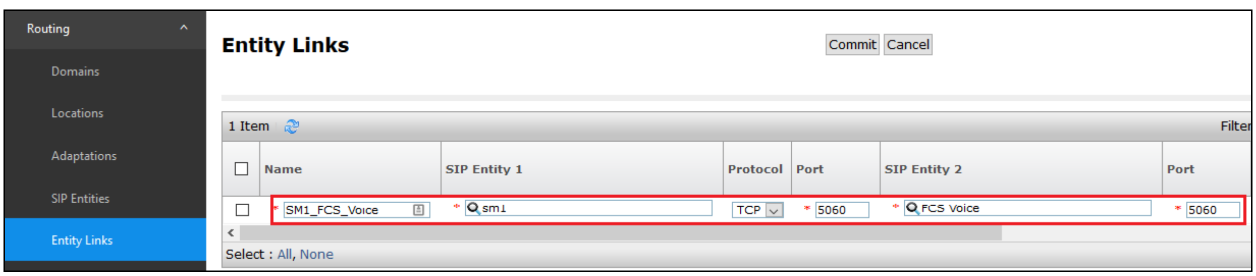
To add an Entity Link, expand **Elements →Routing** and select **Entity Links** from the left navigation menu.

Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link to Session Manager.
- **SIP Entity 1** Select Session Manager already defined.
- **SIP Entity 2** Select the SIP Entity added in **Section 6.3** from drop-down menu.
- **Protocol** After selecting both SIP Entities, verify *TCP* is selected as the required Protocol.
- **Port** Verify **Port** for both SIP entities is *5060*.
- **Connection Policy** Select trusted.

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

The following screen shows the Entity Link defined between FCS Voice Server and Session Manager.



6.5. Define Routing Policy

Routing policies describe the conditions under which calls will be routed.

To add a routing policy, expand **Elements →Routing** and select **Routing Policies**.

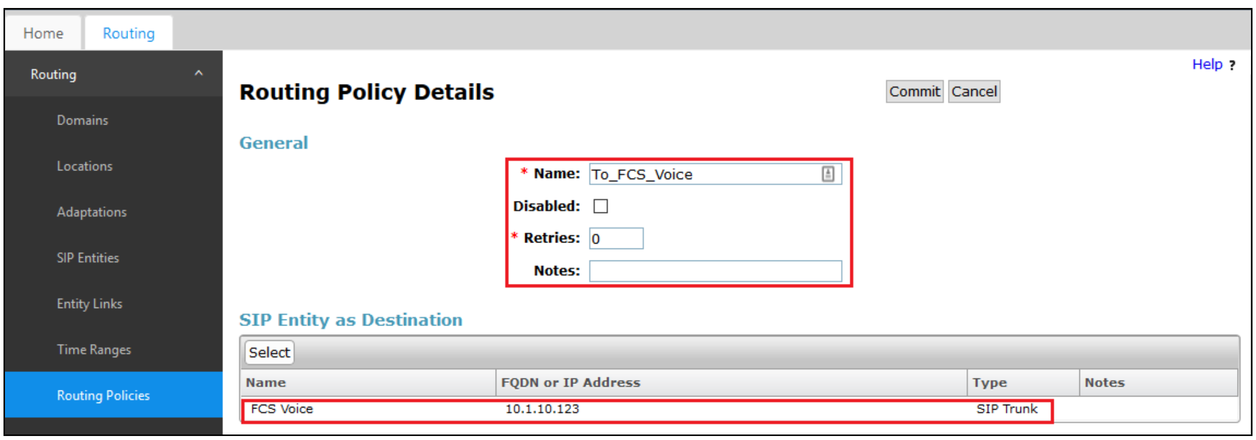
Click **New** (not shown). In the **General** section, enter the following values.

- **Name:** Enter an identifier for routing to FCS Voice Server.
- **Disabled:** Leave unchecked.
- **Retries:** Retain default value of *0*.
- **Notes:** Enter a brief description. [Optional].

In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). Select the SIP Entity defined for FCS Voice in **Section 6.3** and click **Select**.

The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

The following screen shows the Routing Policy for Session Manager.



6.6. Define Dial Pattern

This section describes the steps to define a dial pattern to route calls to FCS Voice Server. In the sample configuration, the Voice Mail Pilot Number 70000 is defined for routing to FCS Voice Server.

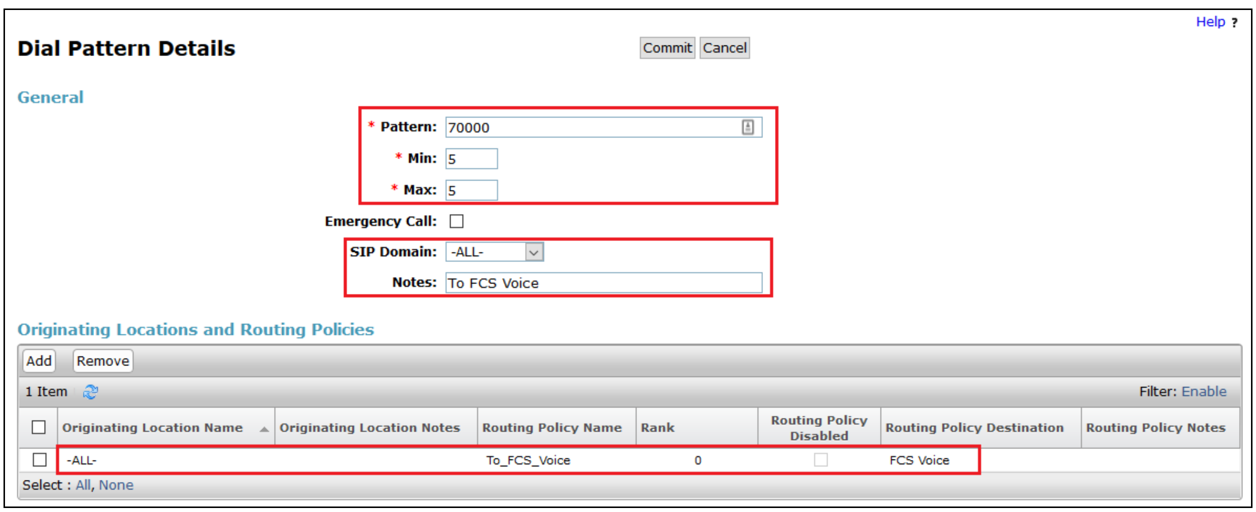
To define a dial pattern, expand **Elements →Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern:** Enter dial pattern for the Voice Mail Pilot number.
- **Min:** Enter the minimum number digits that must be dialed.
- **Max:** Enter the maximum number digits that may be dialed.
- **SIP Domain:** Select the SIP Domain from drop-down menu or select *ALL* if Session Manager should accept incoming calls from all SIP domains.
- **Notes:** Enter a brief description. [Optional].

In the **Originating Locations and Routing Policies** section, click **Add.**
The **Originating Locations and Routing Policy List** page opens (not shown).

- In **Originating Locations** table, select *ALL* .
- In **Routing Policies** table, select the appropriate Routing Policy defined for routing to FCS Voice which is defined in **Section 6.5**.
- Click **Select** to save these changes and return to **Dial Patterns Details** page.

Click **Commit** to save the new definition. The following screen shows the Dial Pattern defined for routing calls to FCS Voice.



5-digit extensions beginning with *71XXX* are assigned to guest rooms are routed to Communication Manager and this is assumed to be defined. Otherwise, Message Waiting Light will not work. SIP NOTIFY messages receive from FCS Voice Server needs to be routed back to Communication Manager.

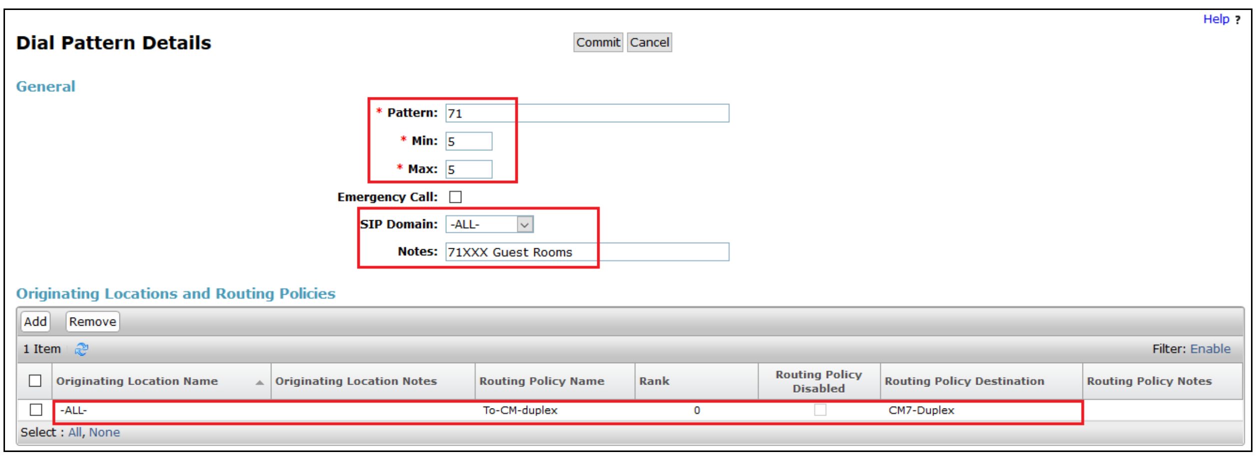
To define a dial pattern, expand **Elements →Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **Pattern:** Enter dial pattern for the guest room numbers, i.e., 71.
- **Min:** Enter the minimum number digits that must be dialed.
- **Max:** Enter the maximum number digits that may be dialed.
- **SIP Domain:** Select the SIP Domain from drop-down menu or select *ALL* if Session Manager should accept incoming calls from all SIP domains.
- **Notes:** Enter a brief description.

In the **Originating Locations and Routing Policies** section, click **Add.**
The **Originating Locations and Routing Policy List** page opens (not shown).

- In **Originating Locations** table, select *ALL* .
- In **Routing Policies** table, select the appropriate Routing Policy defined for routing to Communication Manager which is presumed to be defined in initial setup.
- Click **Select** to save these changes and return to **Dial Patterns Details** page.

Click **Commit** to save the new definition. The following screen shows the Dial Pattern defined for guest rooms.



7. Configure FCS Voice

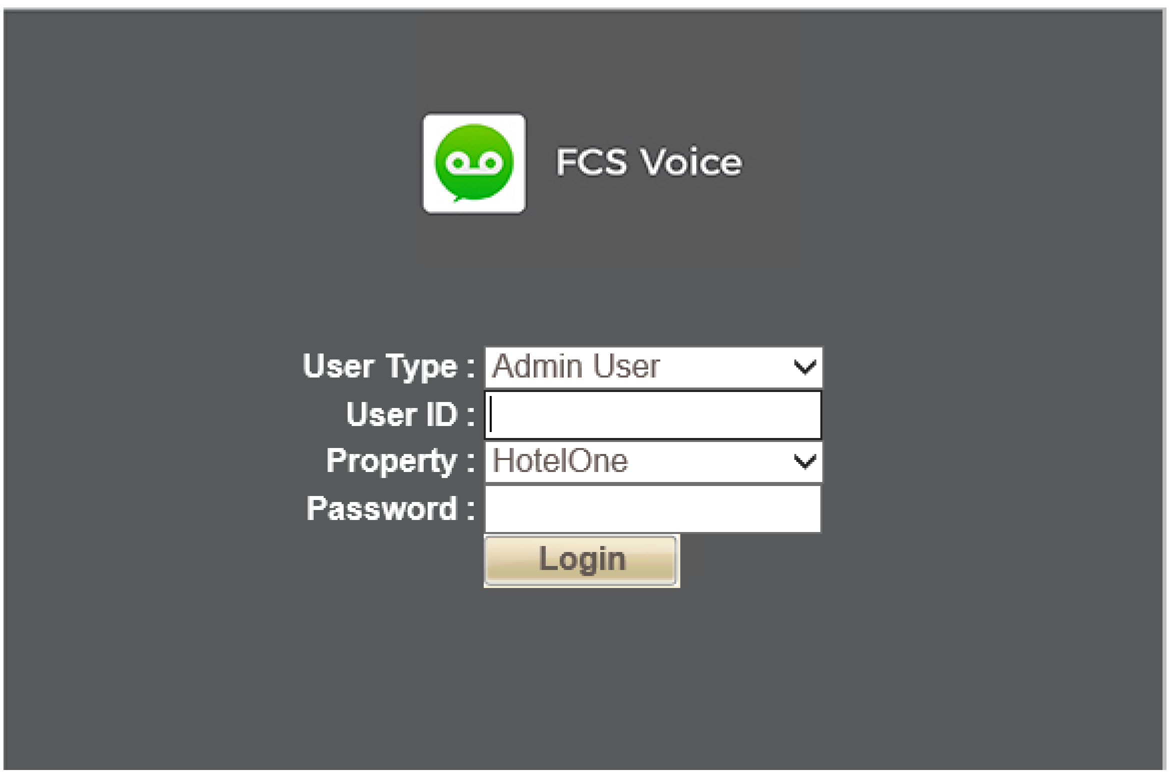
This section details the essential portion of the FCS Voice configuration to interoperate with Communication Manager and Session Manager. These Application Notes assume that the FCS Voice application has already been properly installed by FCS professional services personnel. Further details of the FCS Voice setup can be found in **[7]**.

The following settings will be verified:

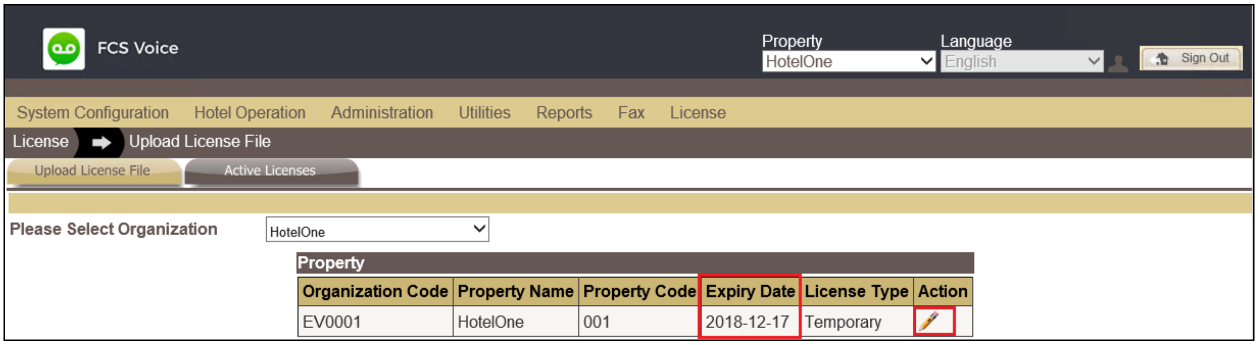
- License Verification
- PBX setting
- SIP Trunking
- Service Numbers

7.1. License Verification

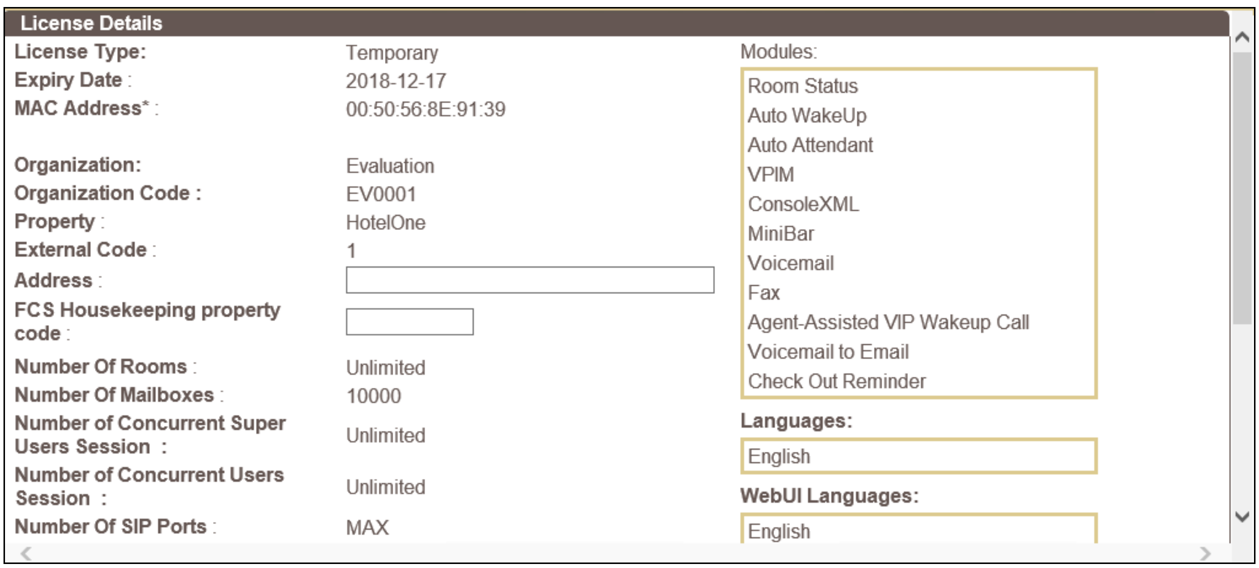
Configuration is accomplished by accessing the browser-based GUI of FCS Voice Server, using the URL **http://localhost/VoicemailWebUI/Login.aspx** on the server**.** Select the appropriate property and log in with the appropriate credentials.



Select from top menu **License → Active Licenses**. Ensure that the License has not expired.

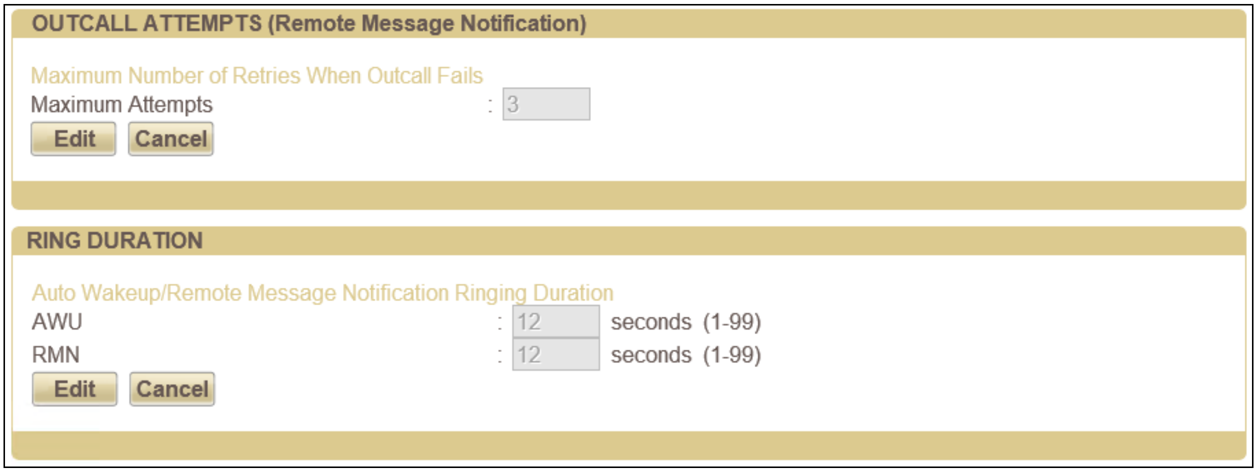


Click on the edit ('pencil') icon under **Action** and view the details. Ensure that the appropriate license parameters are enabled. Note that *Temporary* license was used for this Compliance Testing.



7.2. System Configuration

Select from top menu **System Configuration → System Settings → General Setting**. Verify the Auto Wakeup Outcall Attempts and Ring Duration are suitable for setup of WakeUp service in view of the number of rings for coverage of guest rooms mentioned in **Section 5.8**.

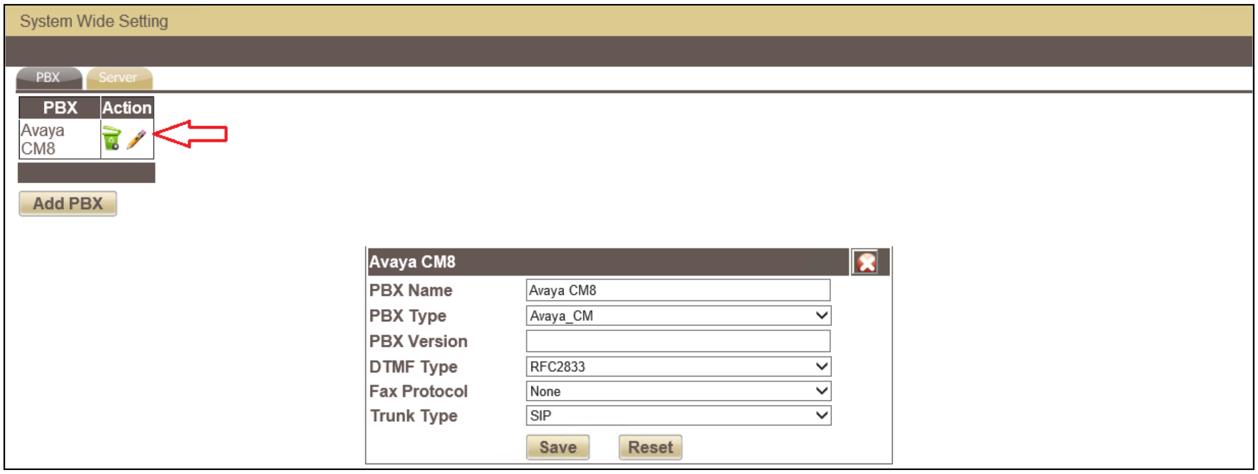


7.3. PBX Setting

From the home screen, select **System Wide Setting** from the drop-down menu.

The following is the resulting screen after log in. Click on the edit ('pencil') icon and view the PBX settings. Ensure that the following settings are configured:

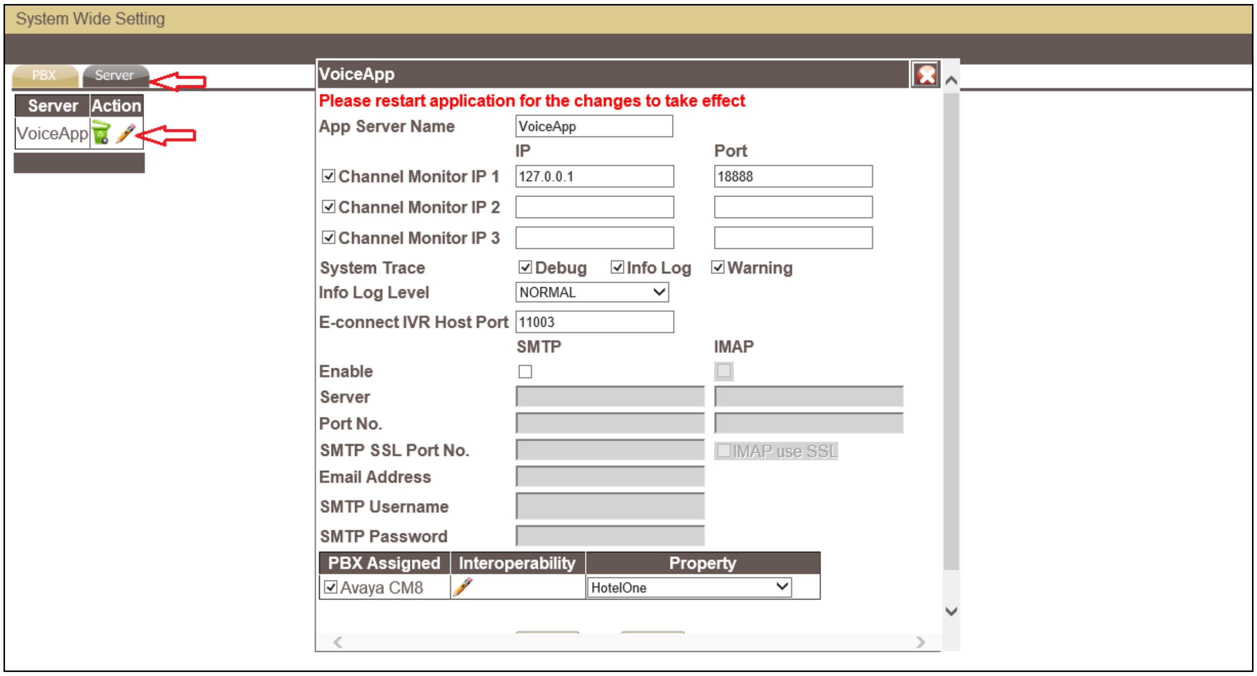
- **PBX Name**: Enter the appropriate name.
- **PBX Type**: Select *Avaya_CM* from the drop-down menu.
- **PBX Version**: Enter appropriate version number. [Optional].
- **DTMF Type**: Select *RFC2833* from the drop-down menu.
- **Trunk Type**: Enter *SIP* for SIP Trunking with Session Manager.



Click **Save** to commit the changes.

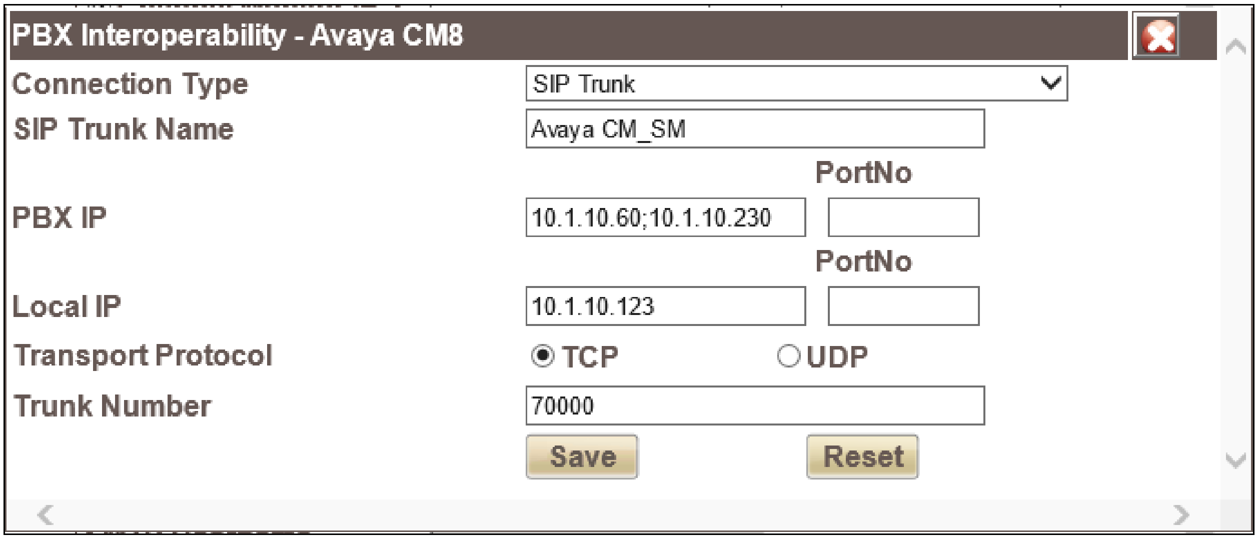
7.4. SIP Trunking

From the System Wide Setting, click on the Server tab on the top left and then the edit ('pencil') icon to show the following Voice Server details. On the checkbox next to the **PBX Assigned** for *Avaya CM8*, click on the edit ('pencil') icon under **Interoperability** below and the next screen shows the SIP Trunking parameters.



The followings are configured for the SIP Trunk:

- **Connection Type:** Select the *SIP Trunk* from drop down menu.
- **SIP Trunk Name:** Enter appropriate name.
- **PBX IP:** Enter Session Manager and Communication Manager IP Addresses (ensure no space between the 2 IPs and separated by semi colon).
- **Local IP:** Enter the FCS Voice Server IP Address.
- **Transport Protocol:** Select *TCP* radio button for communication as defined in Session Manager Entity Link in **Section 6.4**.
- **Trunk Number:** Enter the Voice Mail Pilot Number defined in **Section 5.5.4**.



Click **Save** to commit the changes; click **Save** again on the next screen.

7.5. Service Numbers & Pilot Number

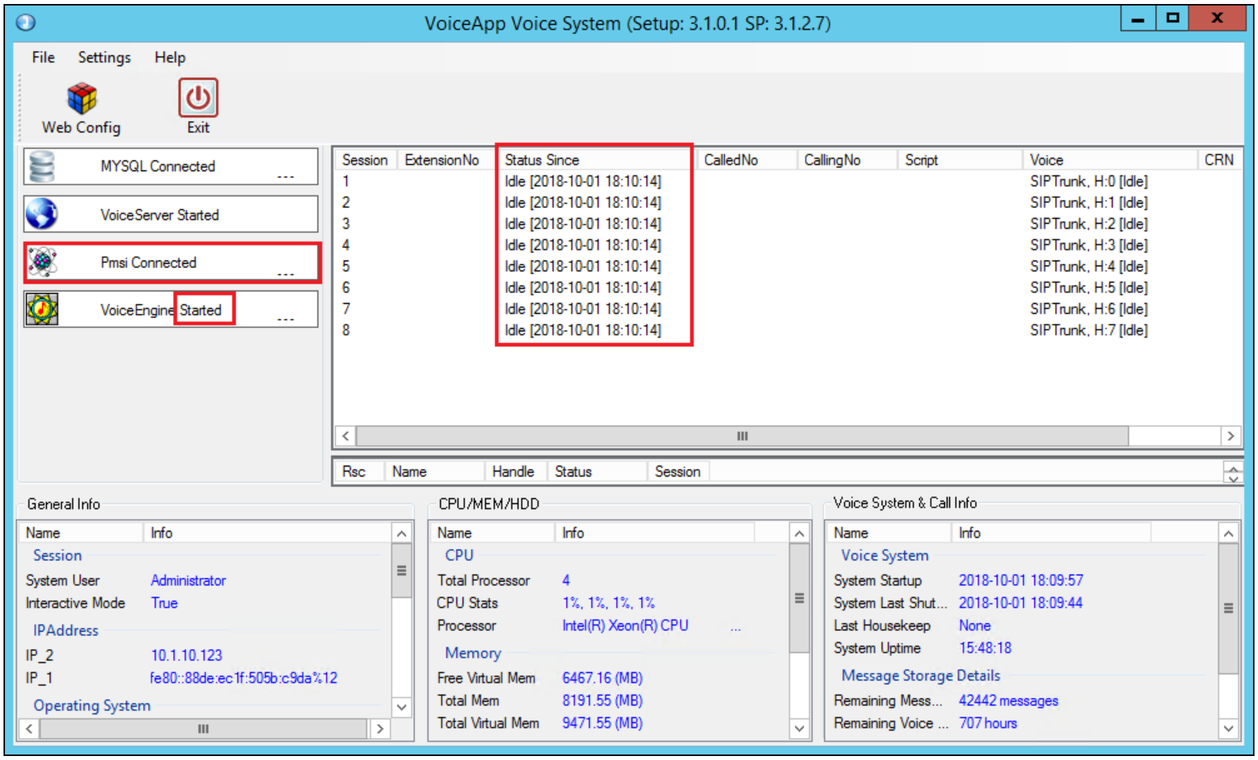
Select **System Configuration → Hardware Settings → Channels → Entry Point** from the home screen. Configure each Service Number (the VDN/Vectors as setup in **Section 5.6** for Configuration of Communication Manager) to a specific service. Map the Pilot Number 70000 to **Direct** Call Flow. Lastly, map **W_W** to **Busy/No Answer** Call Flow.

		Entry Point	CPI Format	Description
		1	70001_W	DIRECT
		2	70002_W	XPRESS MESSAGE LEAVE
		3	70003_W	SETAWU
		4	W_W	BUSY/NOANSWER
		5	70000_W	DIRECT
		1		

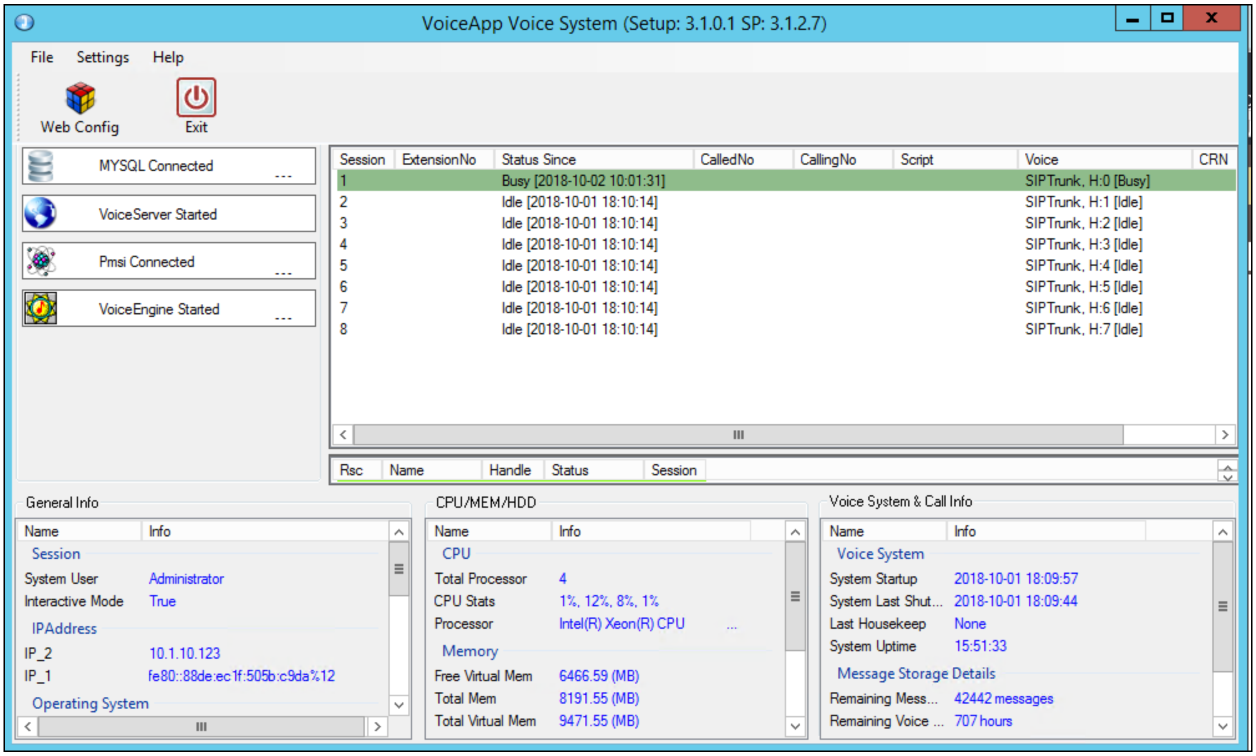
8. Verification Steps

This section describes steps that may be used to verify the configuration. From the FCS Voice Server, launch **FCS Voice** from the Windows Apps select 

. Verify that the **VoiceEngine** status shows *Started* and the voice channels under **Status Since** column are *Idle*. Once the PMS communication has been successfully established, the **PMSI** status will show up as Connected.



Dial the express leave message service number 70002 at one of the admin stations. Observe that one channel of the SIP Trunk is busy as shown below. Verify proper prompt is received and that leaving a voice mail message to either a guest or admin mailbox works.



Check that the message waiting light is turned on. Enter the command **status station x** where **x** is the guest phone number to confirm the **Message Waiting:** indicate *VM Server* and the message waiting light on the deskphone is on. Dial the express message retrieval service number 70001 to retrieve the message. Check that the **Message Waiting:** shows *blank* and the message waiting light on the deskphone is off.

```
status station 71122                                           Page   1 of    7
                              GENERAL STATUS
     Administered Type: 9641G             Service State: in-service/on-hook
        Connected Type: 9641               Signal Status: connected
             Extension: 71122             Network Region: 1
                  Port: S00022        Parameter Download: complete
           Call Parked? no                 SAC Activated? no
      Ring Cut Off Act? no
Active Coverage Option: 1           one-X Server Status: N/A

          EC500 Status: N/A       Off-PBX Service State: N/A
   Message Waiting: VM Server
   Connected Ports:


  Limit Incoming Calls? no

 User Cntrl Restr: none                      HOSPITALITY STATUS
Group Cntrl Restr: none                   Awaken at:
                                           User DND: not activated
                                          Group DND: not activated
                                        Room Status: occupied
```

To verify the Operator transfer function, call any guest room and let it go to coverage on the FCS Voice Server. Press the DTMF digit ‘0’ to select for call to be routed to Operator. Verify call is connected to Operator. Alternatively, set a wakeup call and allow it to ring-out (i.e. do not pick up when it rings) for the maximum number of retries (as pre-configured); after that, the system will call the Operator extension as a form of notification for a wakeup failure.

9. Conclusion

These Application Notes describe the procedures for configuring FCS Voice (SIP) v3.1 to interoperate with Avaya Aura® Communication Manager R8.0 and Avaya Aura® Session Manager R8.0. All interoperability compliance test cases executed against such a configuration were completed successfully.

10. Additional References

The following documents are available at http://support.avaya.com.

[1] *Administering Network Connectivity on Avaya Aura® Communication Manager*, Release 8.0, Issue 1, Aug 2018.

[2] *Administering Avaya Aura® Communication Manager*, Release 8.0, Jul 2018.

[3] *Administering Avaya Aura™ Session Manager*, Release 8.0, Issue 1, Aug 2018.

[4] *Deploying Avaya Aura® Session Manager in Virtual Appliance*, Release 8.0, Issue 2, Sep 2018.

[5] *Application Notes for FCS Gateway with Avaya Aura® Communication Manager R8.0.*

The following documents are provided by FCS Computer Systems Sdn Bhd.

[6] *FCS Voice v3.1 Configuration Manual, Version 3.4, 29 Jun 2018.*

[7] *FCS Voice v3.1 Installation Manual (Windows Server 2012 R2), Version 3.5, 26 Sep 2018.*

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Developer*Connection* Program at devconnect@avaya.com.