Avaya Solution & Interoperability Test Lab

Application Notes for FCS Voice (SIP) v3.1 with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1 - 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 R10.1 and Avaya Aura® Session Manager R10.1. FCS Voice is an 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 R10.1 and Avaya Aura® Session Manager R10.1. FCS Voice (SIP) v3.1 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 (e.g., 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 functions.

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 Communication Manager with an Avaya G430 Media Gateway, System Manager and Session Manager, and an FCS Voice server including PMS simulator. Avaya Session Border Controller for Enterprise was used to complete a SIP trunk connection to simulate a PSTN connection to the enterprise solution. 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 to Session Manager which acts as proxy to Communication Manager.

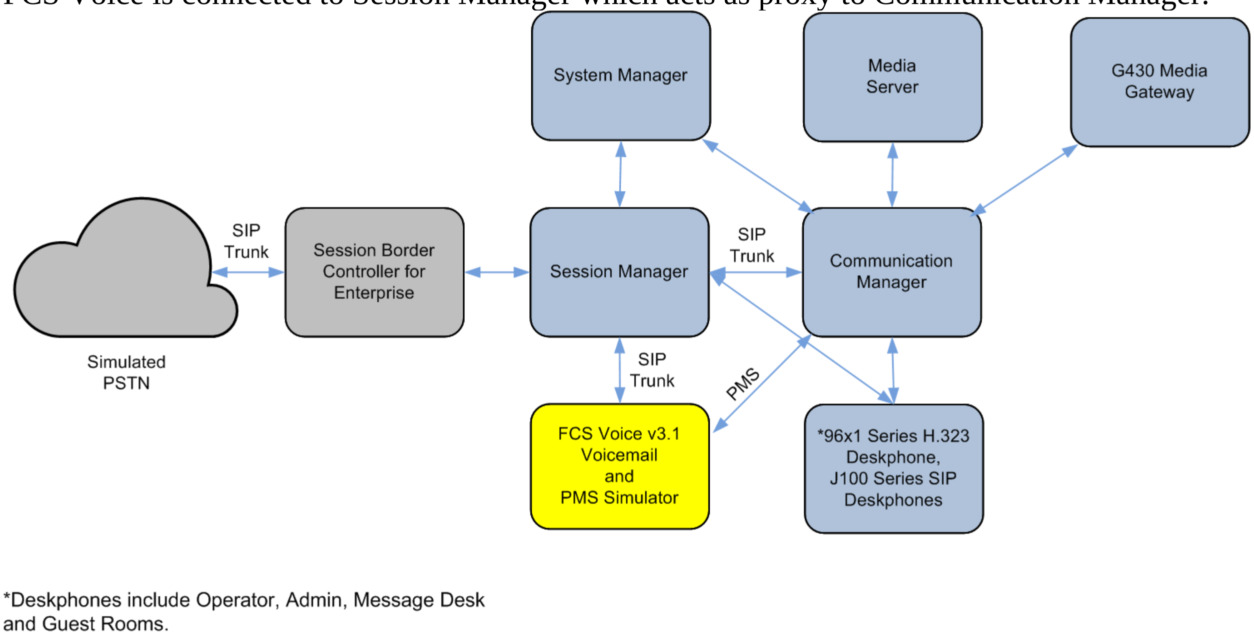


Figure 1: Sample Test Configuration

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	10.1 (10.1.0.0.0.974.27372)
Avaya Aura® Media Server	10.1.0.77
Avaya G430 Media Gateway	42.4.0
Avaya Aura® Session Manager	10.1 SP2 (10.1.0.2.1010215)
Avaya Aura® System Manager	10.1 SP2 Build 10.1.0.0.537353 Hot Fix 1010215160
Avaya Session Border Controller	10.1.0.0.-32-21432
Avaya J100 Series SIP Telephones Avaya J100 Series H.323 Telephones Avaya 96X1 H.323 Deskphones Avaya 14XX Digital Telephones Analog Phones	4.0.11.0 6.8532 6.8532 2.0 SP9 (R20) -
Voice V3.1 Unicorn/PMS Simulator	3.1.0.1

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 **[**111**]** and **[**222222**]** 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 enabled on **Page 2** and **Page 5**:

- **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
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     1
                   Maximum Administered SIP Trunks: 40000     38
  Max Administered Ad-hoc Video Conferencing Ports: 24000      0
   Max Number of DS1 Boards with Echo Cancellation: 999    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? n
  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 **list node-names v4** and note entry for Session Manager name (in this case, *sm1*) and the corresponding **IP Address** (in this example, *10.1.10.60*).

```
list node-names v4

                        NODE NAMES

Type     Name              IP Address
IP       sm1               10.1.10.60
IP       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-group**, 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 configuration of the SIP trunk for calls to Session Manager, which routes calls FCS Voice. It includes the following:

- Configure IP Codec
- Check SIP Trunk Group
- Administer Uniform Dialplan
- Set Private Numbering
- Administer Routing for Voice Mail Calls

5.5.1. Configure IP Codec

IP Network Region 6 is used for calls from Communication Manager to Session Manager. Enter **change ip-codec-set 6** and setup the appropriate codec acceptable by FCS Voice. In this example, *G.711Mu* and *G.711A* audio codecs are administered for IP Network Region 6 assigned for calls to FCS Voice Server. Leave the rest as default. Codec *G.729* was also tested but not defined below.

```
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. Check SIP Trunk-Group

Enter **change trunk n,** where **n** is the number of the SIP trunk group to Session Manager (in this example, trunk-group **7**). Ensure the following parameter is set:

- **Numbering Format:** Enter *private*
- **Support Request History:** Enter *y*
- **Telephone Event Payload Type:** Enter *101*

```
change trunk-group 7                                              Page   3 of   4
 TRUNK FEATURES
          ACA Assignment? n            Measured: both
                                                        Maintenance Tests? y



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

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n


                                 Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

```
change trunk-group 7                                              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? 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
     Resend Display UPDATE Once on Receipt of 481 Response? 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
```

5.5.3. Administer Uniform Dialplan

The Voice Mail Pilot Number 70000 is set up on FCS Voice in **Section** 7.4. This needs to be part of uniform dialing to dial the number without AAR access code. Enter **change uniform-dialplan 7** to configure the uniform dial plan for 70000. At the **Matching Pattern** *70000*, enter the **Len** as *5* and the **Net** as *aar* for dialing through 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
```

5.5.4. Set 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
```

5.5.5. Administer Routing for Voice Mail Calls

Enter **change aar analysis x** for routing calls to Voice Mail Pilot Number 70000 to FCS Voice. 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 existing SIP trunk group number under the column **Grp No** which is *7***. Numbering Format** is set as *lev0-pvt* to set private numbering for calling number to FCS Voice.

```
change route-pattern 6                                          Page   1 of   4
                    Pattern Number: 6      Pattern Name: non-IMS 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: 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 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
```

5.6. 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 FCS Voice WebUI.*

The corresponding settings on FCS Voice are detailed in **Section 7.5**.

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*:

SIP URI:

* 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 3

                          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. Create 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* for the aar access code.

```
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









SIP URI:
```

```
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. Create 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   5
                                     STATION

Extension: 71121                         Lock Messages? n               BCC: 0
     Type: 9611                          Security Code: *                TN: 1
     Port: S000192                     Coverage Path 1: 70              COR: 5
     Name: William                     Coverage Path 2:                 COS: 5
Unicode Name? n                     Hunt-to Station:                  Tests? y
STATION OPTIONS
                                           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              Button Modules: 0
 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

                                              Customizable Labels? y
```

```
change station 71121                                            Page   2 of   5
                                     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? y
                                              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 1                                       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 Communication Manager and Session Manager, and defining Communication Manager as a Managed Element. Please refer to **[**3**3]** and **[Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**Error! Not a valid bookmark self-reference.**4**]** for additional details.

The following administration activities will be described:

- 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 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.

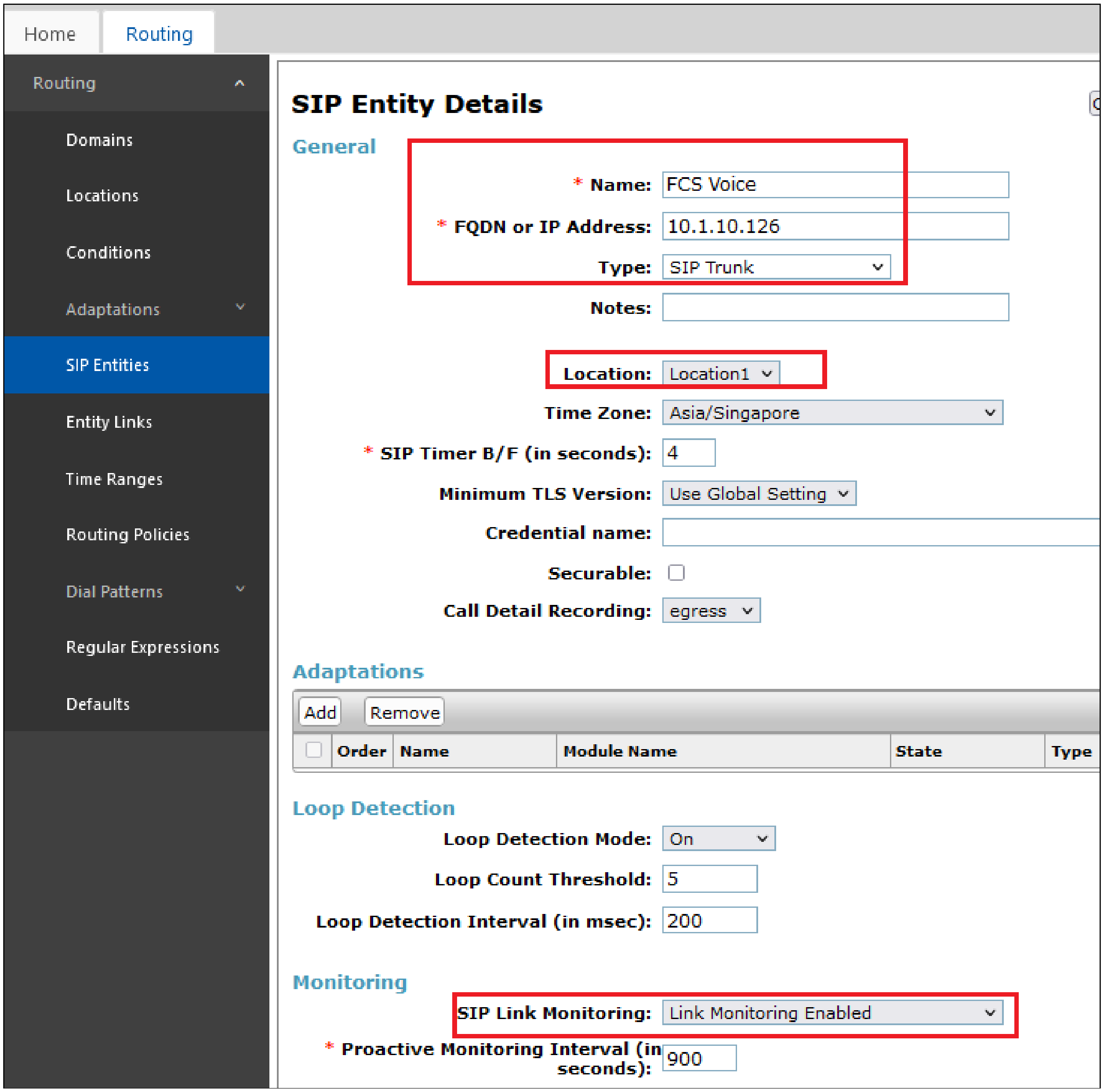
Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

- **Name:** Enter an identifier for new SIP Entity.
  In the sample configuration, *FCS Voice* was used.
- **FQDN or IP Address:** Enter IP address as *10.1.10.126.*
- **Type:** Select *SIP Trunk.*
- **Notes:** Enter a brief description. [Optional].
- **Location:** Select appropriate Location defined for Communication Manager.

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.



Home
Routing
Routing
Domains
Locations
Conditions
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
SIP Entity Details
General
* Name: FCS Voice
* FQDN or IP Address: 10.1.10.126
Type: SIP Trunk
Notes:
Location: Location1
Time Zone: Asia/Singapore
* SIP Timer B/F (in seconds): 4
Minimum TLS Version: Use Global Setting
Credential name:
Securable:
Call Detail Recording: egress
Adaptations
Add
Remove
Order
Name
Module Name
State
Type
Loop Detection
Loop Detection Mode: On
Loop Count Threshold: 5
Loop Detection Interval (in msec): 200
Monitoring
SIP Link Monitoring: Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds): 900

6.2. 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 Link were added between Session Manager and FCS Voice Server.

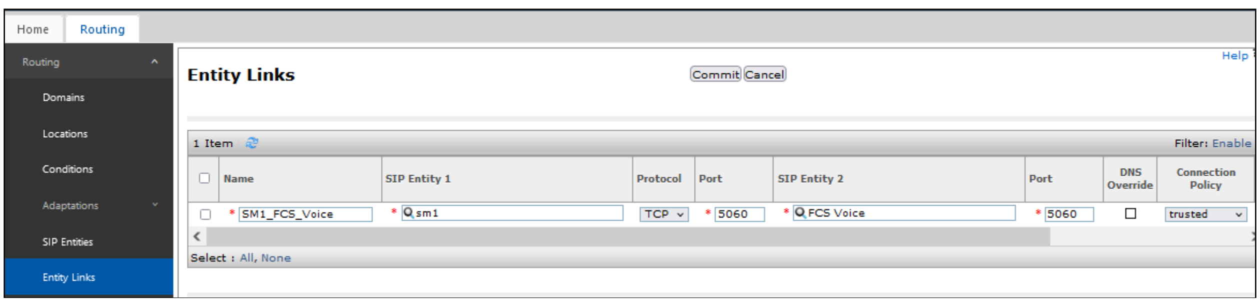
To add an Entity Link, expand **Elements** → **Routing** and select **Entity Links** from the left navigation menu.

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 6.1** 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.



6.3. 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**.

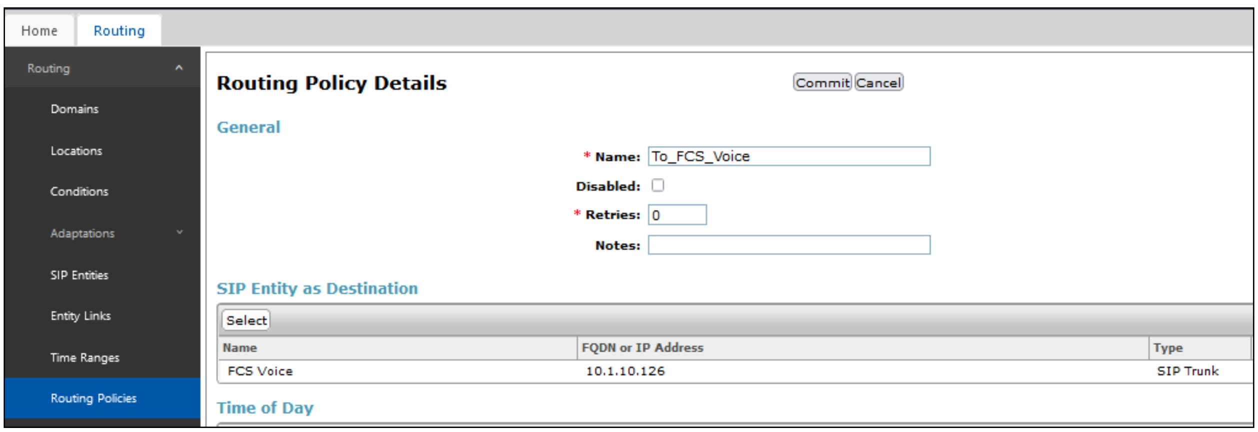
Click **New** (not shown). In the **General** section, enter the following values.

- **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 6.1** 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.



6.4. 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.

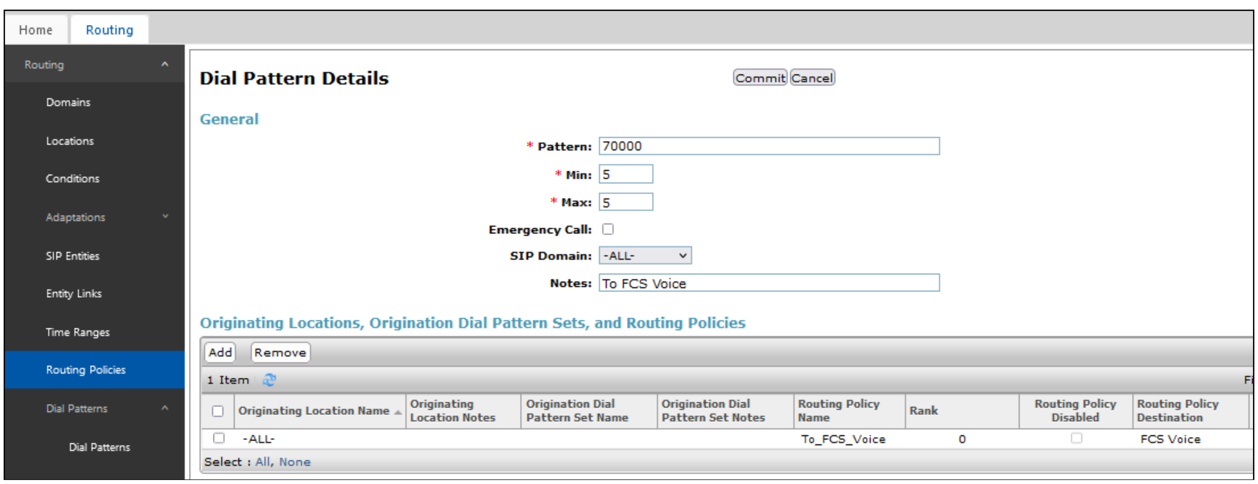
To define a dial pattern, expand **Elements →Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **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, Origination Dial Pattern Sets, and Routing Policies** section, click **Add.** The **Originating Locations, Origination Dial Pattern Sets, 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 6.3**.
- 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.

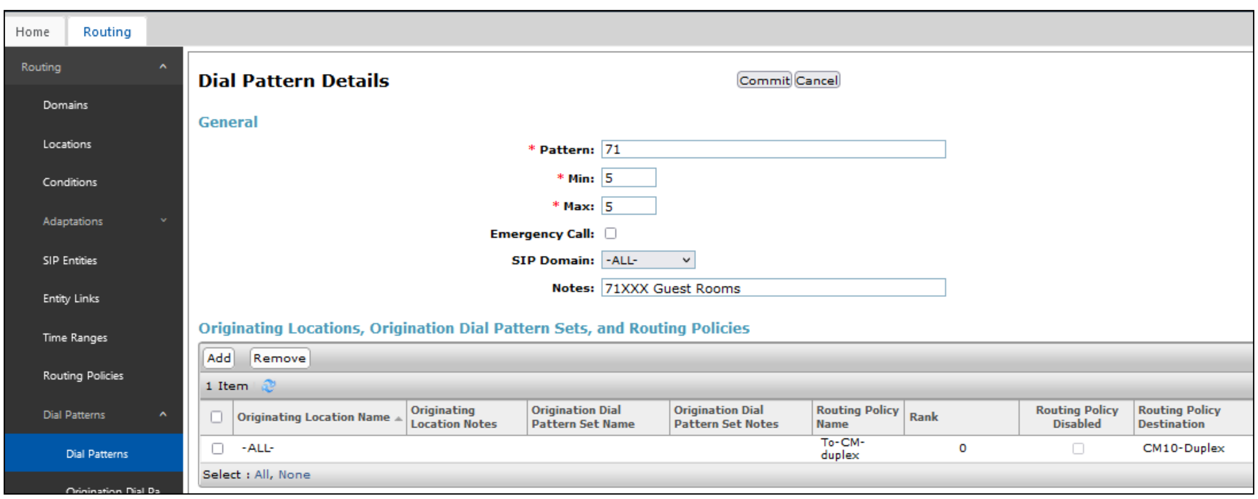
To define a dial pattern, expand **Elements** → **Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

- **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, Origination Dial Pattern Sets, and Routing Policies** section, click **Add.** The **Originating Locations, Origination Dial Pattern Sets, 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

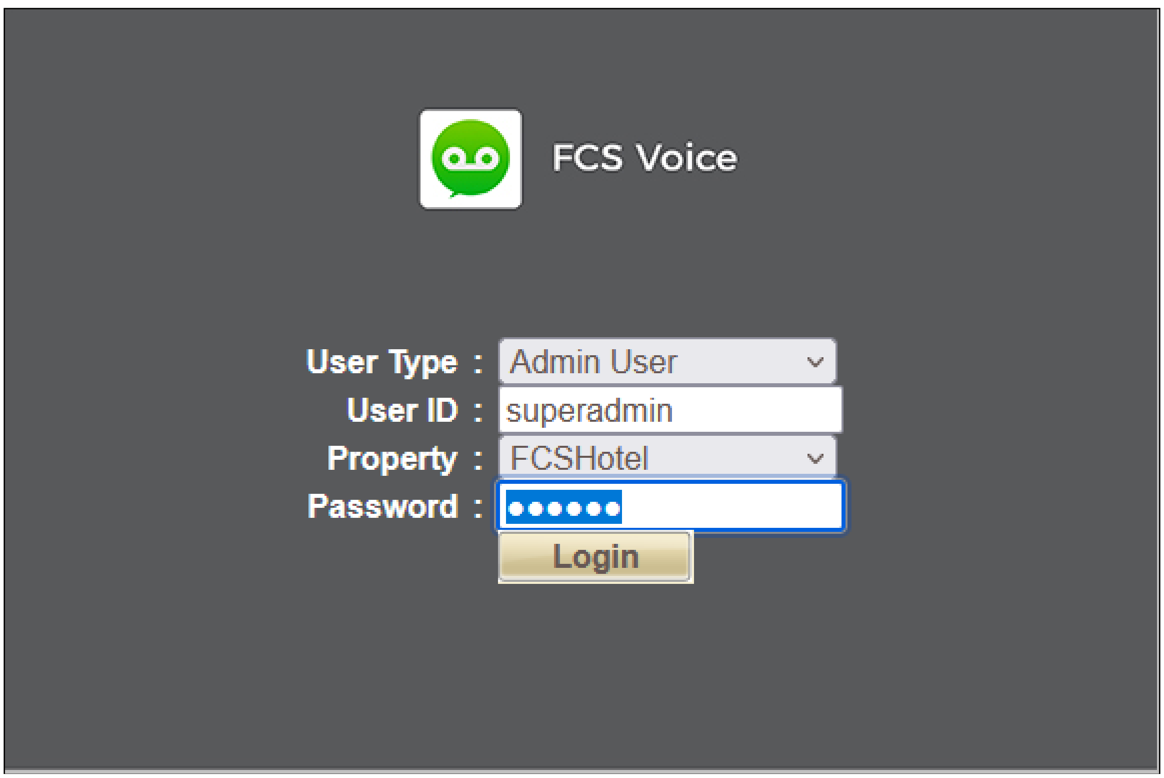
This section details the essential portion of the FCS Voice configuration to interoperate with Communication Manager and Session Manager. These Application Notes assume that the FCS Voice application has already been properly installed by FCS professional services personnel. Further details of the FCS Voice setup can be found in **[77]**.

The following settings will be verified:

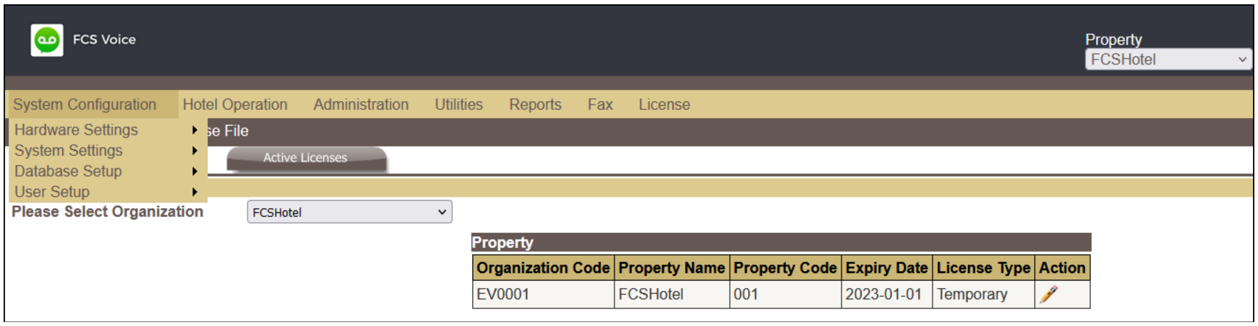
- License Verification
- PBX Setting
- SIP Trunking
- Service Numbers

7.1. License Verification

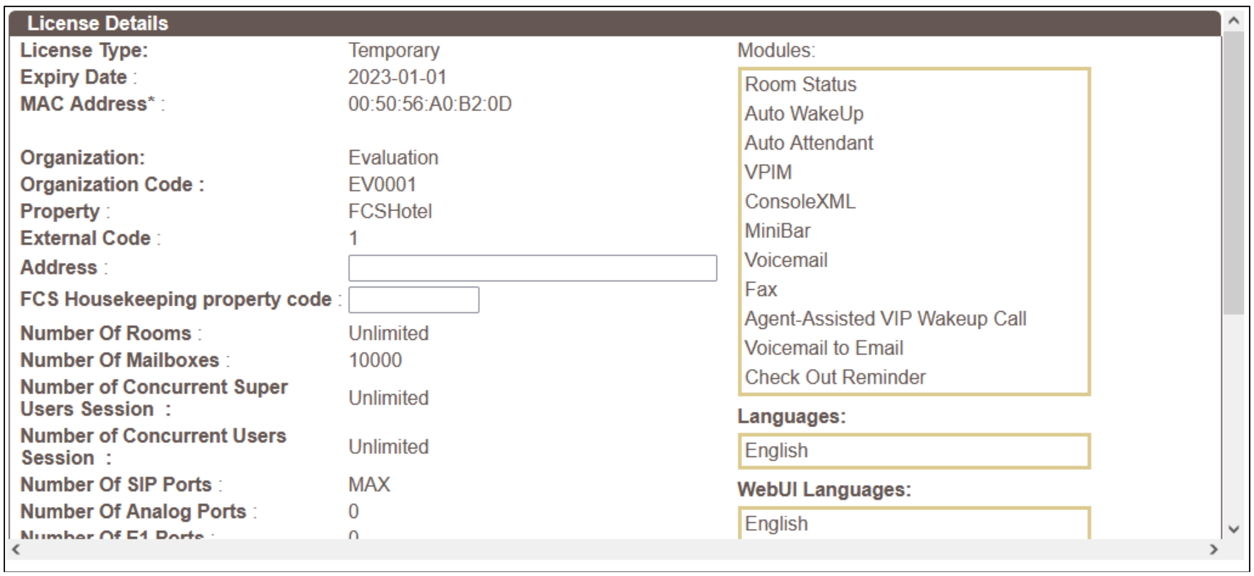
Configuration is accomplished by accessing the browser-based GUI of FCS Voice Server, using the URL **http://localhost/VoicemailWebUI/Login.aspx** on the server**.** Select the appropriate property and log in with the appropriate credentials.



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

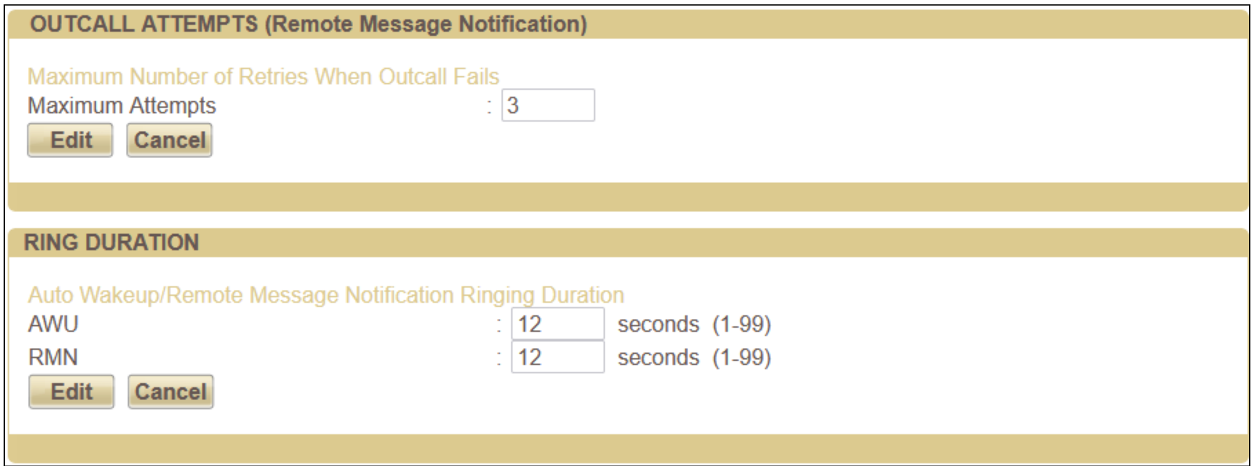


Click on the edit ('pencil') icon under **Action** and view the details. Ensure that the appropriate license parameters are enabled. Note that *Temporary* license was used for this compliance testing.



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**.

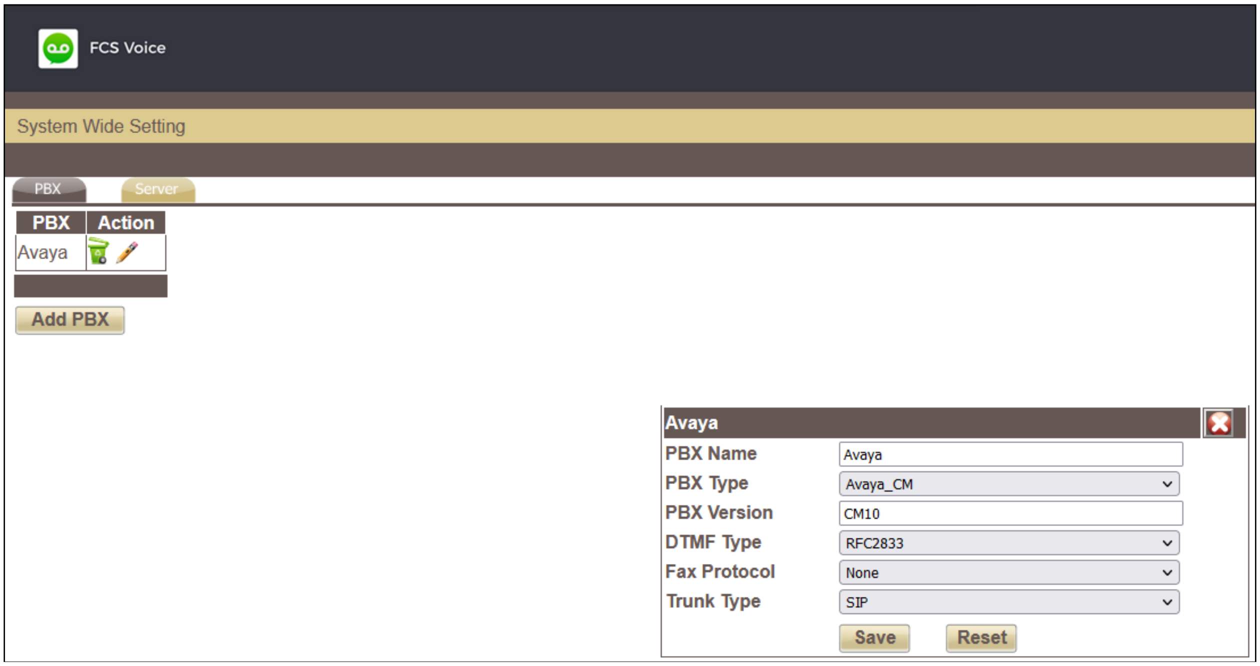


7.3. PBX Setting

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:

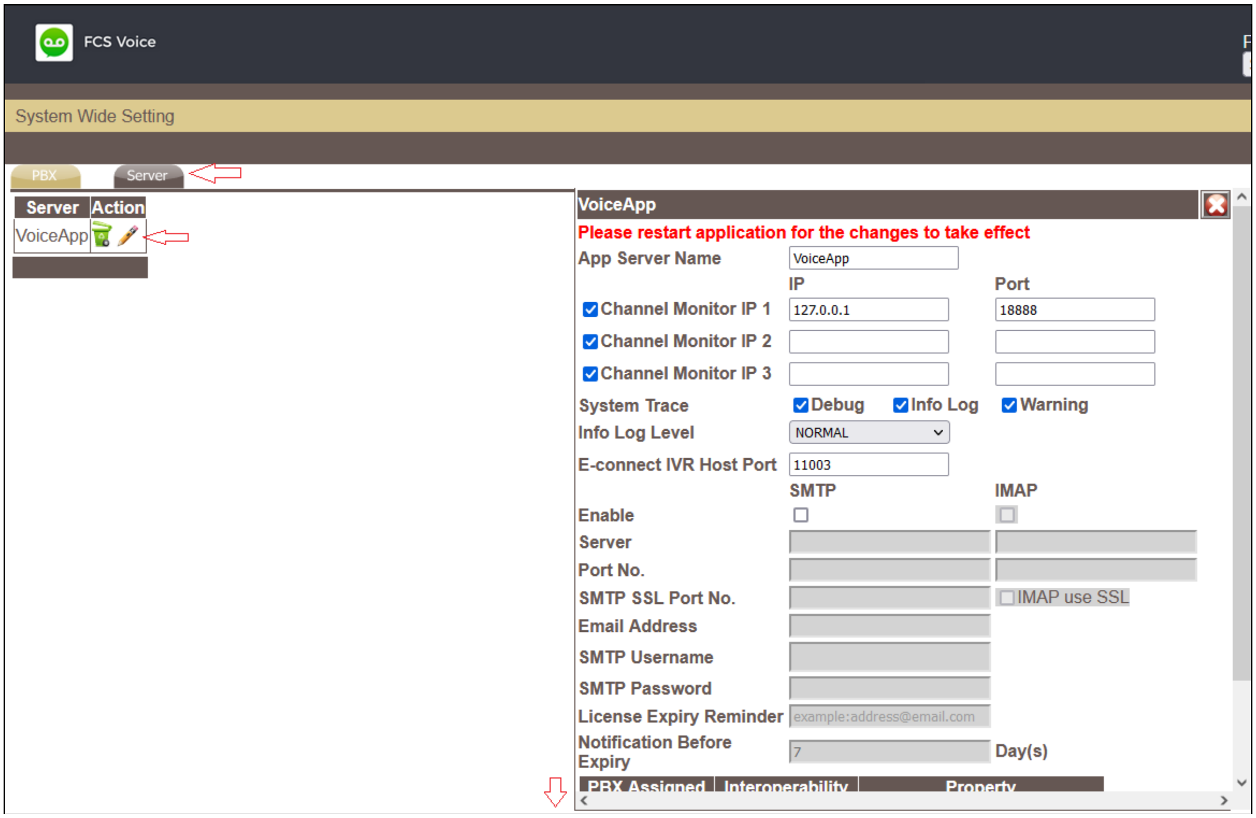
- **PBX Name**: Enter the appropriate name.
- **PBX Type**: Select *Avaya_CM* from the drop-down menu.
- **PBX Version**: Enter appropriate version number (optional).
- **DTMF Type**: Select *RFC2833* from the drop-down menu.
- **Trunk Type**: Enter *SIP* for SIP Trunking with Session Manager.

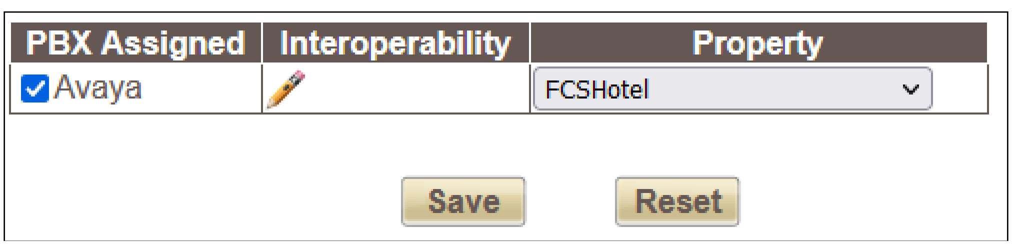


Click **Save** to commit the changes.

7.4. SIP Trunking

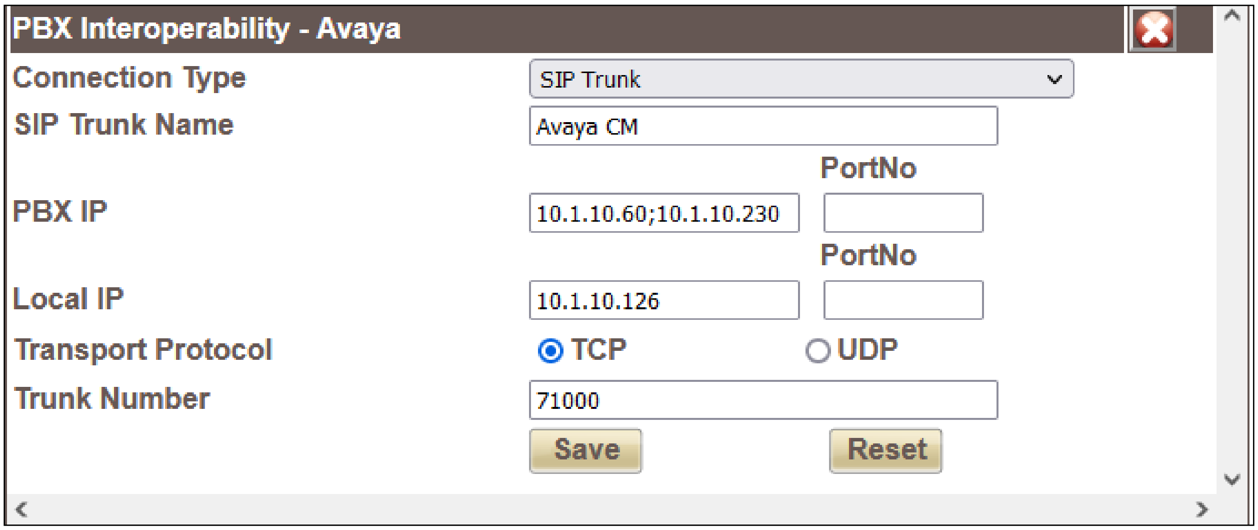
From **System Wide Setting**, click on the **Server** tab on the top left and then the edit ('pencil') icon to show the following Voice Server details. On the checkbox next to the **PBX Assigned** for *Avaya* below, click on the edit ('pencil') icon under **Interoperability.** The next screen shows the 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 6.2**.
- **Trunk Number:** This setting was not utilized in the integration with Session Manager.



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

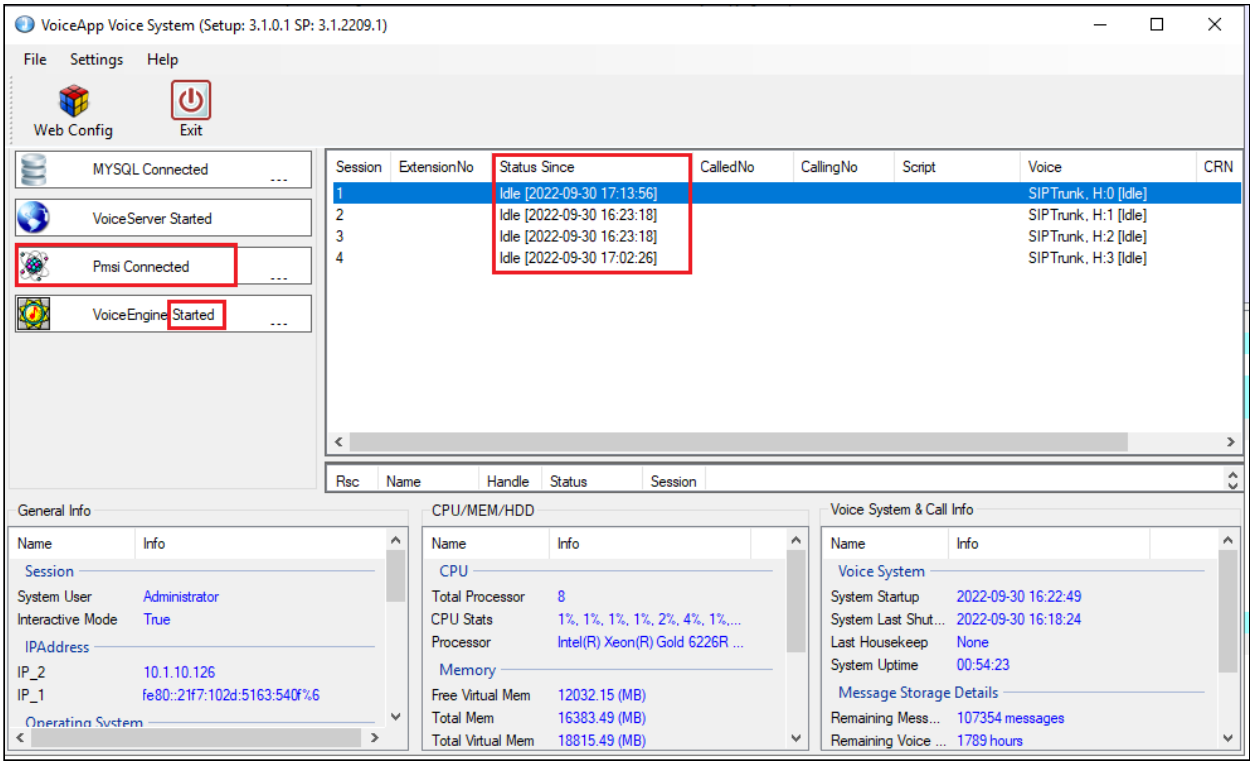
7.5. Service Numbers & Pilot Number

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.

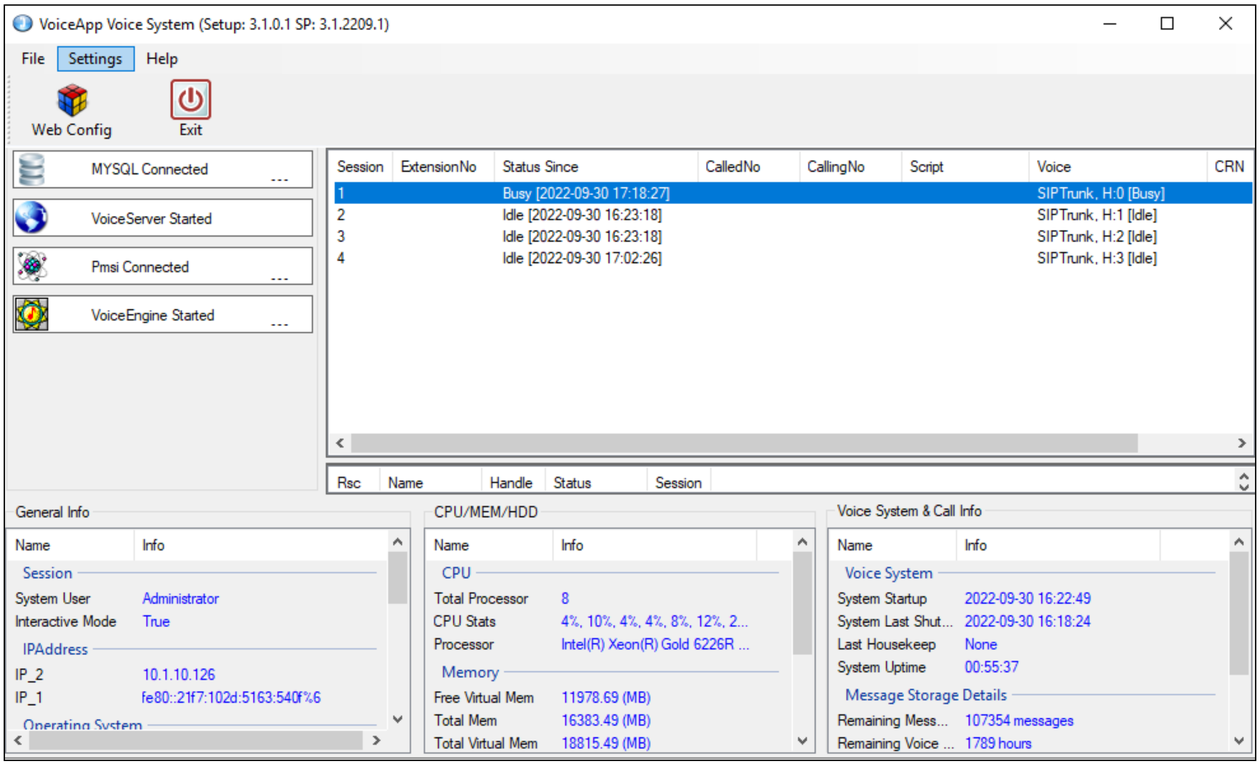
		Entry Point	CPI Format	Description
		1	W_W	BUSY/NOANSWER
		2	70001_W	DIRECT
		3	70002_W	XPRESS MESSAGE LEAVE
		4	70003_W	SETAWU
		5	70000_W	DIRECT
		1		

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** indicates *VM Server PMS* 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: 9611G                Service State: in-service/on-hook
        Connected Type: 9611                 Signal Status: connected
             Extension: 71122               Network Region: 1
                  Port: S000022         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  PMS
   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 R10.1 and Avaya Aura® Session Manager R10.1. 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 10.1.x, Issue 1, Sep 2022

[2] *Administering Avaya Aura® Communication Manager,* Release 10.1, Issue 1, Dec 20121.

[3] *Administering Avaya Aura™ Session Manager*, Release 10.1, Issue 4, Sep 2022.

[**Error! Not a valid bookmark self-reference.**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 R10.1*.

The following documents are provided by FCS Computer Systems.

[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.