Avaya Solution & Interoperability Test Lab

Application Notes for Telematrix 3302IP and 9602IP SIP telephones with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Telematrix 3302IP and 9602IP SIP telephones which were compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify Telematrix 3302IP and 9602IP SIP telephones functionalities in an environment that comprised of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and various Avaya H.323 and SIP IP telephones.

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 Telematrix 3302IP and 9602IP SIP telephones which were compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services. Telematrix 3302IP and 9602IP SIP telephones are SIP based IP telephones that deliver superior performance for small to midsize conference rooms.

The Telematrix 3302IP and 9602IP Cordless DECT are SIP-only hospitality guest sets. As such, they are configurable at the factory as 5-button speed dial or 10-button speed dial sets. While they are capable of "Conference", they are not configured for "Transfer" as that is an undesirable trait in hotel guest room environments. The Phones are powered only by POE/802.3af and both devices draw approximately 3.5W which classify them as Class 2 POE Devices. Both phone types will acquire an IP address when met by a DHCP server.

These Application Notes assume that Avaya Aura Communication Manager and Avaya Aura SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [1][2][3].

1.1. Interoperability Compliance Testing

The overall objective of the interoperability compliance testing is to verify Telematrix 3302IP and 9602IP SIP telephones functionalities in an environment that comprised of Avaya Aura Communication Manager, Avaya SIP Enablement Services, various Avaya IP Telephones, and various Avaya SIP endpoints.

The interoperability compliance test included verification of features and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on Telematrix endpoints. Telematrix endpoints operations such as inbound calls, outbound calls, hold, forward, conference, Feature Name Extension (FNE), and Telematrix endpoints interaction with Avaya SIP Enablement Services (SES), Avaya Aura Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. The serviceability testing introduced failure scenarios to see if Telematrix endpoints can recover from failures.

1.2. Support

Technical support for Telematrix 3302IP and 9602IP SIP telephones can be obtained by contacting Telematrix via the support link at http://sipsupport.telematrix.net or by calling 800-462-9446.

2. Introduction

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G450 Media Gateway, an Avaya SIP Enablement Services (SES) server, and Telematrix 3302IP and 9602IP SIP telephones. During the compliance test, an Avaya S8300 server with a G450 Media Gateway was utilized, since the system contained the internal audix system for MWI testing. The solution described herein is also extensible to other Avaya Media Servers and

Media Gateways. Avaya S8720 Servers with an Avaya G650 Media Gateway were included in the test to provide inter-switch call scenarios. For completeness, Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, Avaya 9600 Series H.323 IP Telephones, and Avaya 6400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based Telematrix 3302IP and 9602IP SIP telephones and Avaya SIP, H.323, and digital telephones.

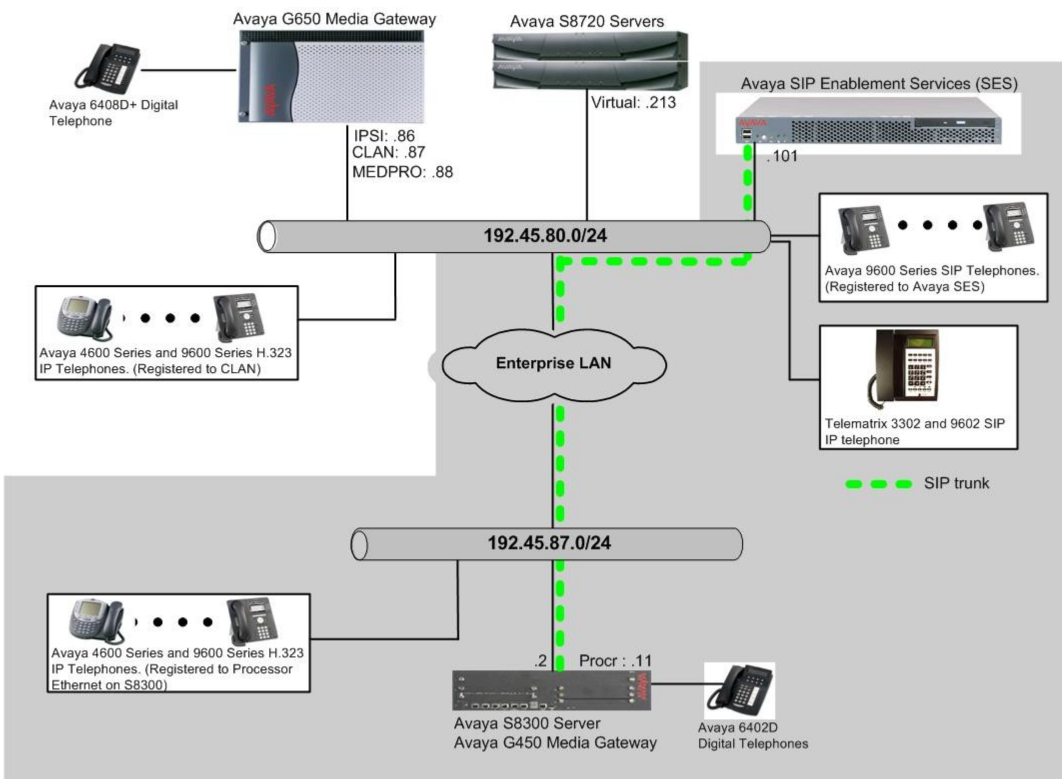


Figure 1: Test Configuration of Telematrix 3302IP and 9602IP SIP telephones

3. Equipment and Software Validated

The following equipment and software were used for the test configuration.

Equipment		Software
Avaya S8720 Servers		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G650 Media Gateway		
	TN2312BP IPSI TN799DP CLAN TN2302AP MEDPRO	HW11 FW030 HW20 FW017 HW01 FW108
Avaya S8300 Server		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G450 Media Gateway		28.17
Avaya SIP Enablement Services		Avaya Aura™ SIP Enablement Services 5.2 (R015x.02.0.947.3) with Service Pack SES-02.0.947.3-SP2a
Avaya 4600 Series IP Telephone		
	4620SW 4625SW	2.9 2.9
Avaya 9600 Series IP Telephone		
	9630 9650	3.002 3.002
Avaya 9600 Series SIP Telephone		
	9620 9630	2.05 2.05
Avaya 64xx Series Digital Telephones		
	6408D+	-
Telematrix		
	3302IP 9602IP	1.8.82-595 1.8.90-603

4. Configure Avaya Aura Communication Manager

This section describes the procedure for setting up a SIP trunk between Avaya Aura Communication Manager and Avaya Aura SES. The steps include setting up a list of IP codecs in an IP codec set, an IP network region, an IP node name, a signaling group, a trunk group, and a SIP station. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

These steps are performed from the Avaya Aura Communication Manager System Access Terminal (SAT) interface. Telematrix 3302IP and 9602IP SIP telephones and other SIP telephones are configured as off-PBX telephones in Avaya Aura Communication Manager.

4.1. Capacity Verification

Enter the **display system-parameters customer-options** command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 22                            RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 900    52
                                      Maximum Stations: 450    9
                              Maximum XMOBILE Stations: 0      0
                    Maximum Off-PBX Telephones - EC500: 50     0
                    Maximum Off-PBX Telephones -   OPS: 100    5
                    Maximum Off-PBX Telephones - PBFMC: 0      0
                    Maximum Off-PBX Telephones - PVFMC: 0      0
                    Maximum Off-PBX Telephones - SCCAN: 0      0
```

On **Page 2** of the form, verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 100    33
           Maximum Concurrently Registered IP Stations: 450    3
             Maximum Administered Remote Office Trunks: 0      0
Maximum Concurrently Registered Remote Office Stations: 0      0
              Maximum Concurrently Registered IP eCons: 0      0
  Max Concur Registered Unauthenticated H.323 Stations: 0      0
                   Maximum Video Capable H.323 Stations: 5      0
                    Maximum Video Capable IP Softphones: 5      0
                       Maximum Administered SIP Trunks: 100    10
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
   Maximum Number of DS1 Boards with Echo Cancellation: 0      0
                             Maximum TN2501 VAL Boards: 0      0
                     Maximum Media Gateway VAL Sources: 0      0
           Maximum TN2602 Boards with 80 VoIP Channels: 0      0
          Maximum TN2602 Boards with 320 VoIP Channels: 0      0
   Maximum Number of Expanded Meet-me Conference Ports: 0      0
```

4.2. IP Codec Set

This section describes the steps for administering a codec set in Avaya Aura Communication Manager. This codec set is used in the IP network region for communications between Avaya Aura Communication Manager and Avaya Aura SES. Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 4.3** for configuring IP network regions to specify which codec sets may be used within and between network regions. For the compliance testing, G.711MU and G.729 were tested for verification.

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

                          IP Codec Set
    Codec Set: 1

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

To configure a specific codec for Avaya 9600 Series SIP phones, the **46xxsettings.txt** file must be configured. The following shows the **CODEC SETTINGS** section that need to be modified, accordingly.

```
.
.
.
#####################  CODEC SETTINGS  #####################
##
## G.711a Codec Enabled
##   Determines whether G.711 a-law codec is available on
##   the phone.
##     0 for No
##     1 for Yes
## SET ENABLE_G711A 1 (This shows the default)
##
##Added the following statement:
SET ENABLE_G711A 0 (Modified G.711A codec to disable)
##
## G.711u Codec Enabled
##   Determines whether G.711 mu-law codec is available on
##   the phone.
##     0 for No
##     1 for Yes
## SET ENABLE_G711U 1 (This shows the default)
##
##Added the following statment:
SET ENABLE_G711u 0 (Modified G.711U_codec to disable)
##
## G.729 Codec Enabled
##   Determines whether G.729 codec is available on the
##   phone.
##     0 for G.729(A) disabled
##     1 for G.729(A) enabled without Annex B support
##     2 for G.729(A) enabled with Annex B support
## SET ENABLE_G729 1
##e
##
## G.726 Codec Enabled
##   Determines whether G.726 codec is available on the
##   phone.  This parameter is not supported on 16cc phones.
##     0 for No
##     1 for Yes
## SET ENABLE_G726 1
```

```
##
## G.726 Payload Type
##   Specifies the RTP payload type to be used with the
##   G.726 codec. (96-127).  This parameter is not supported
##   on 16cc phones.
## SET G726_PAYLOAD_TYPE 110
##
## G.722 Codec Enabled
##   Determines whether G.722 codec is available on the
##   phone.  This parameter is not supported on 16cc phones.
##     0 for No
##     1 for Yes
## SET ENABLE_G722 0
SET ENABLE_G722 1
##
## DTMF Payload Type
##   Specifies the RTP payload type to be used for RFC
##   2833 signaling. (96-127).
## SET DTMF_PAYLOAD_TYPE 120
##
## DTMF Transmission Method
##   Specifies whether DTMF tones are sent in-band, as
##   regular audio, or out-of-band, using RFC 2833
##   procedures.
##     1 for in-band
##     2 for out-of-band using RFC 2833
## SET SEND_DTMF_TYPE 2
##
.
.
.
```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Avaya Aura Communication Manager for communication between Avaya Aura Communication Manager and Avaya Aura SES. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- Authoritative Domain – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to **moris.mot.com**. This should match the SIP Domain value on Avaya Aura SES, in **Section 5.1**.
- Intra-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Aura Communication Manager or Avaya Aura SES in the same IP network region. The default value for this field is **yes**.
- Codec Set – Set the codec set number as provisioned in **Section 4.2**.
- Inter-region IP-IP Direct Audio – Set to **yes** to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Aura Communication Manager or Avaya Aura SES in different IP network regions. The default value for this field is **yes**.

```
change ip-network-region 1                                    Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: moris.mot.com
    Name:
```

```
MEDIA PARAMETERS                    Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                  Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                          IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                      RTCP Reporting Enabled? y
 Call Control PHB Value: 46          RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46           Use Default Server Parameters? y
        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
```

4.4. Configure IP Node Name

This section describes the steps for setting an IP node name for Avaya Aura SES in Avaya Aura Communication Manager. Enter the **change node-names ip** command, and add a node name for Avaya Aura SES along with its IP address.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
CLAN                    192.45.80.87
IA770                   192.45.87.12
SES                     192.45.80.101
default                 0.0.0.0
procr                   192.45.87.11
```

4.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Avaya Aura Communication Manager for communication between Avaya Aura Communication Manager and Avaya SIP Enablement Services. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- Group Type – Set to **sip.**
- Near-end Node Name - Set to **procr** as displayed in **Section 4.4**.
- Far-end Node Name - Set to the Avaya Aura SES name configured in **Section 4.4**.
- Far-end Network Region - Set to the region configured in **Section 4.3**.
- Far-end Domain - Set to **moris.mot.com**. This should match the SIP Domain value in **Section 5.1**.

```
add signaling-group 3                                         Page   1 of   1

 Group Number: 3              Group Type: sip
                        Transport Method: tls
```

```
  Near-end Node Name: procr                  Far-end Node Name: SES
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
      Far-end Domain: moris.mot.com

                                           Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                      IP Audio Hairpinning? n
        Enable Layer 3 Test? n
 Session Establishment Timer(min): 3            Alternate Route Timer(sec): 6
```

4.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Avaya Aura Communication Manager for communication between Avaya Aura Communication Manager and Avaya Aura SES. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- Group Type – Set the Group Type field to **sip**.
- Group Name – Enter a descriptive name.
- TAC (Trunk Access Code) – Set to any available trunk access code.
- Signaling Group – Set to the Group Number field value configured in **Section 4.5**.
- Number of Members – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Note: *Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted*

```
add trunk-group 3                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: ToSES                       COR: 1        TN: 1        TAC: 1003
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                         Signaling Group: 3
                                                       Number of Members: 10
```

4.7. Configure SIP Endpoint

This section describes the steps for administering OPS stations in Avaya Aura Communication Manager and associating the OPS station extensions with the telephone numbers of Telematrix 3302IP and 9602IP SIP telephones. Enter **add station s**, where **s** is an extension valid in the provisioned dial plan. The following fields were configured for the compliance test.

- Type – Set to **9600SIP**.
- Name – Enter a descriptive name

Repeat this step as necessary to configure additional SIP endpoint extensions.

```
add station 27003                                              Page   1 of   5
                                     STATION

Extension: 27003                       Lock Messages? n               BCC: 0
     Type: 9600SIP                     Security Code:                  TN: 1
     Port: IP                        Coverage Path 1:                 COR: 1
     Name: SIP 27003                 Coverage Path 2:                 COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 2          Personalized Ringing Pattern: 1
             Data Module? n                Message Lamp Ext: 27003
            Speakerphone: 2-way         Mute Button Enabled? y
        Display Language: english

          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n
```

Enter the **add off-pbx-telephone station-mapping** command and configure the following:

- Station Extension – Set the extension of the OPS station as configured above.
- Application – Set to **OPS**.
- Phone Number – Enter the number that Telematrix 3302IP and 9602IP SIP telephones will use for registration and call termination. In the example below, the Phone Number is the same as the Station Extension, but is not required to be the same.
- Trunk Selection – Set to the trunk group number configured in **Section 4.6**.
- Config Set – Set to **1**.

Repeat this step as necessary to configure additional off-pbx-telephone station-mapping.

```
add off-pbx-telephone station-mapping                           Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station         Application Dial   CC  Phone Number       Trunk        Config
 Extension                   Prefix                        Selection    Set
 27003              OPS           -     27003                3            1
```

The following Avaya feature name extension (FNE) set was utilized during the compliance test. Enter **display off-pbx-telephone feature-name-extensions set 1** to view the feature name extensions. The highlighted fields are tested during the compliance test.

```
display off-pbx-telephone feature-name-extensions set 1         Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

        Active Appearance Select: 27051
             Automatic Call Back: 27052
      Automatic Call-Back Cancel: 27053
                Call Forward All: 27054
     Call Forward Busy/No Answer: 27055
             Call Forward Cancel: 27056
                       Call Park: 27057
           Call Park Answer Back: 27058
                    Call Pick-Up: 27059
            Calling Number Block: 27060
          Calling Number Unblock: 27061
  Conditional Call Extend Enable:
 Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer: 27062
           Directed Call Pick-Up: 27063
           Drop Last Added Party: 27064
```

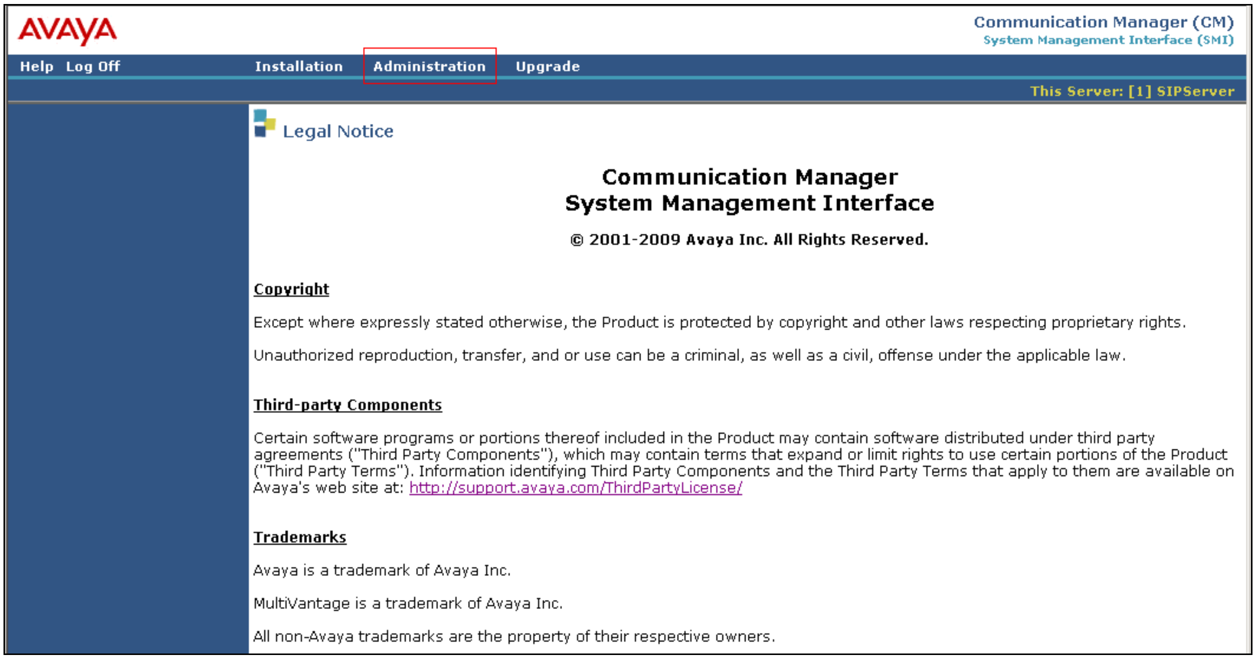
```
display off-pbx-telephone feature-name-extensions set 1         Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off): 27065
      Extended Group Call Pickup: 27066
          Held Appearance Select: 27067
          Idle Appearance Select: 27068
              Last Number Dialed: 27069
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable: 27072
            Off-Pbx Call Disable: 27073
                   Priority Call: 27074
                          Recall:
                  Send All Calls: 27075
           Send All Calls Cancel: 27076
               Transfer Complete:
             Transfer On Hang-Up: 27077
          Transfer to Voice Mail: 27078
         Whisper Page Activation: 27079
```

5. Configure Avaya SIP Enablement Services

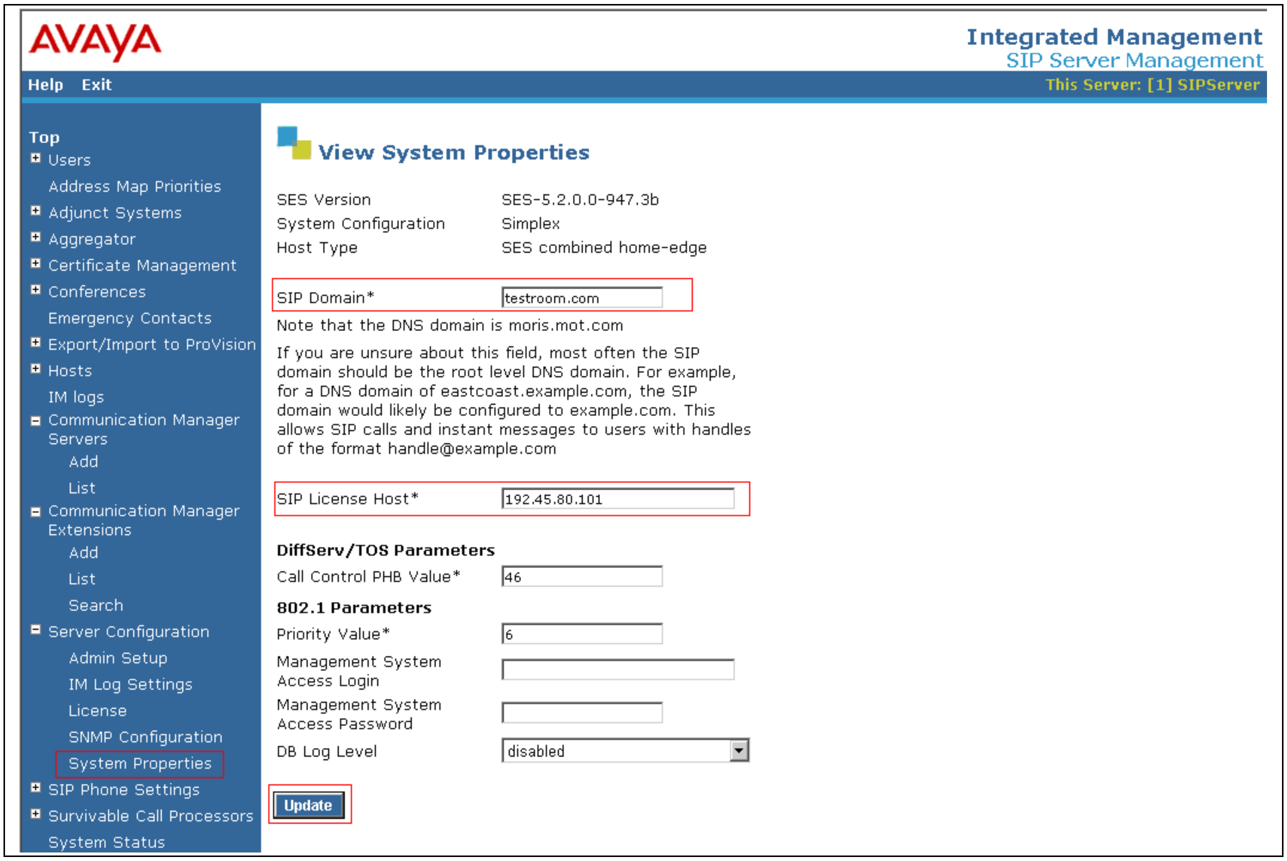
This section describes the steps for creating SIP trunk between Avaya Aura SES and Avaya Aura Communication Manager. SIP user accounts are configured in Avaya Aura SES and associated with an Avaya Aura Communication Manager OPS station extension. Telematrix 3302IP and 9602IP SIP telephones will register with Avaya Aura SES using the SIP user accounts. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

5.1. Configure SES Server Properties

Launch a web browser, enter **https://<IP address of SES server>/admin** in the URL, and log in with the appropriate credentials. Click on the **Launch SES Administration Interface** link upon successful login. Navigate to **Administration → SIP Enablement Services**.



In the Integrated Management SIP Server Management page, select the **Server Configuration → System properties** link from the left pane of the screen. Verify the SIP Domain matches the Far-end Domain field value configured for the signaling group on Avaya Aura Communication Manager in **Section 4.3** and **Section 4.5**. Click on the **Update** button, after the completion.

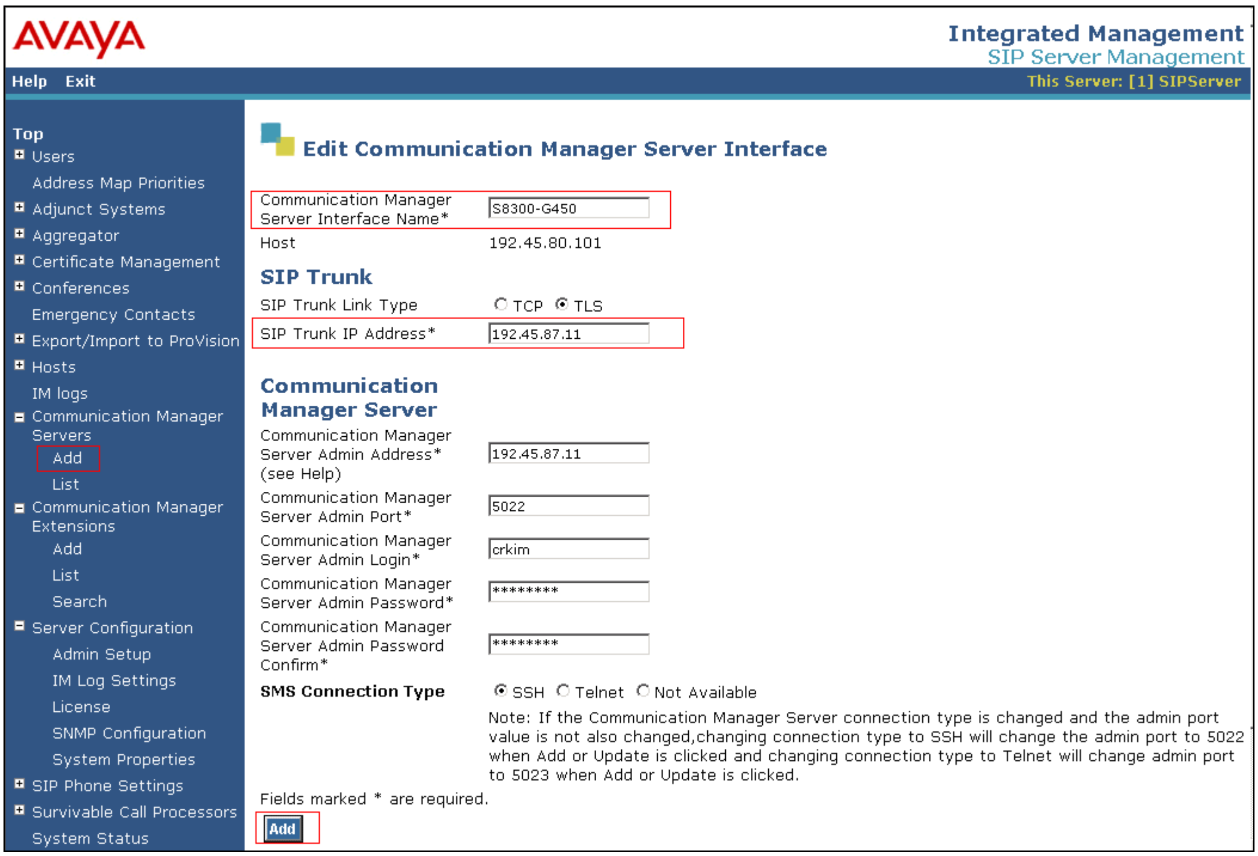


5.2. Configure Communication Manager Server Interface

This section provides steps to add SIP-enabled media servers to the SIP domain. In the Integrated Management SIP Server Management page, select the **Communication Manager Servers → Add** link from the left pane of the screen. The following screen shows the Edit Media Server Interface page. The highlighted fields were configured for the compliance test:

- Communication Manager Server Interface Name – Enter a descriptive name for the communication manager server interface.
- SIP Trunk IP Address – Enter the IP address for the media server's procr (or CLAN) IP interface that terminates the SIP link from SES.

Click **Add** when finished.

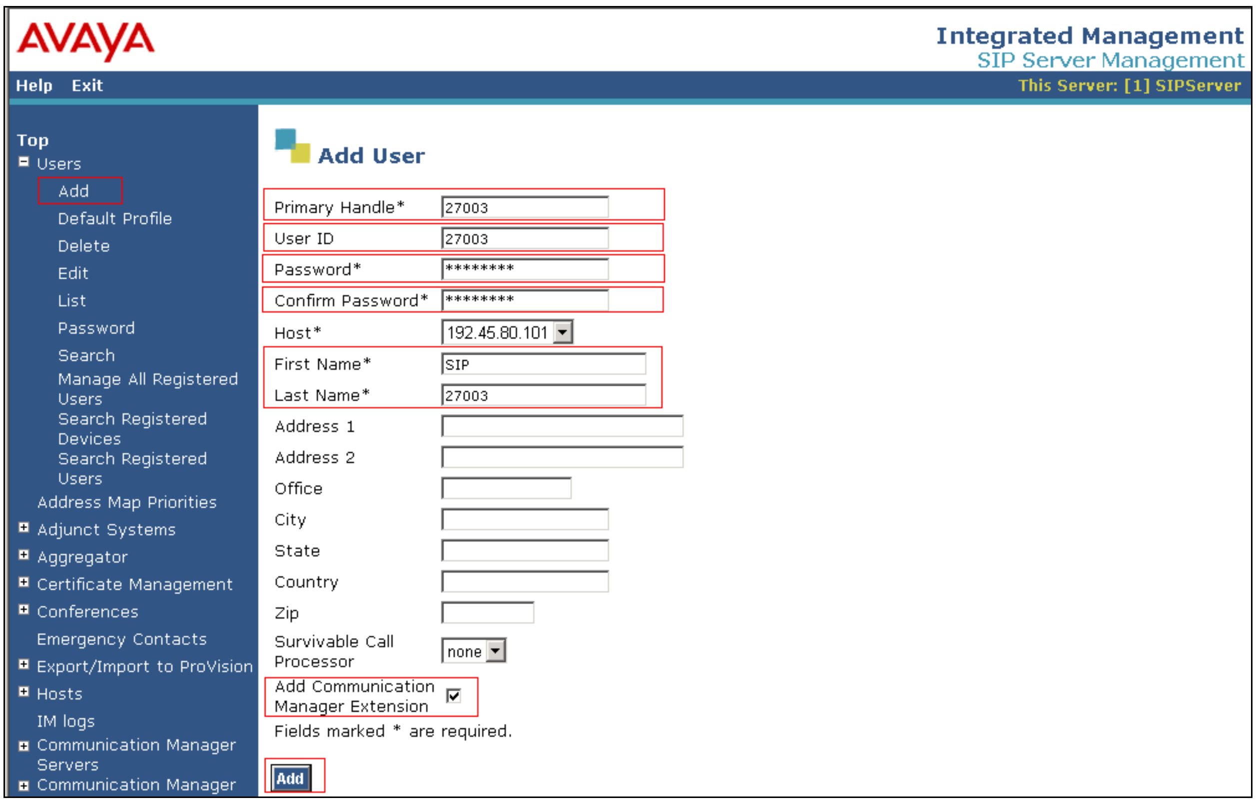


5.3. Configure Users

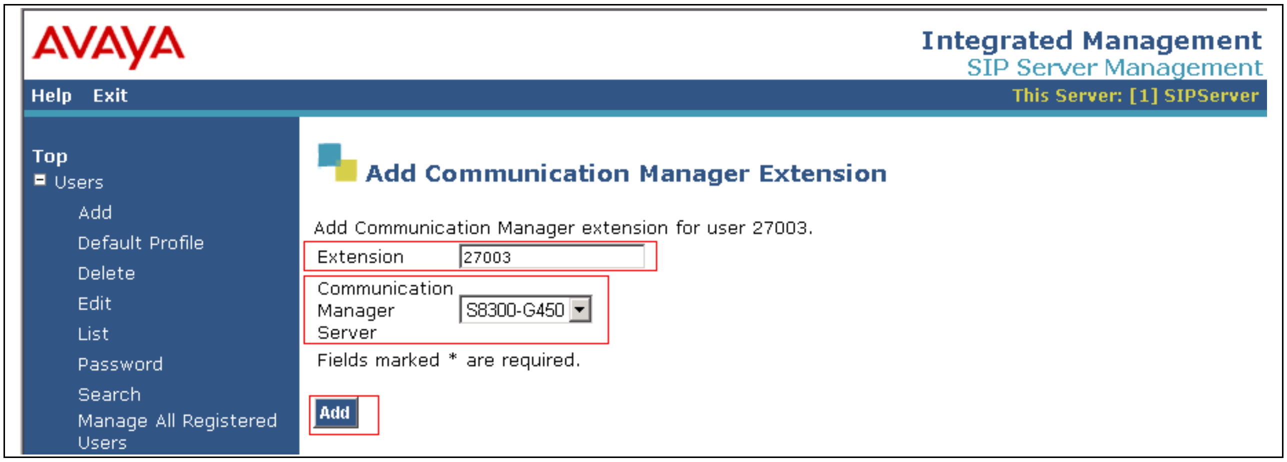
This section provides steps to add users to be administered in the SIP Enablement Services (SES) database. In the Integrated Management SIP Server Management page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test

- Primary Handle – Enter the phone number of Telematrix 3302IP and 9602IP SIP telephones. This number was configured in **Section 4.7**.
- User ID – The same Primary Handle extension was used for this field.
- Password / Confirm Password – Enter a password; both field entries must match exactly.
- First Name – Enter the first name of the user in alphanumeric characters.
- Last Name – Enter the last name of the user in alphanumeric characters.
- Add Media Server Extension - Select this field if you want to associate a new extension number with this user in the database now. If so, the Add Communication Manager Extension screen will be displayed next, after this user profile has been added. If not, in the future you may choose to associate extensions with the user.

Click **Add** when finished.

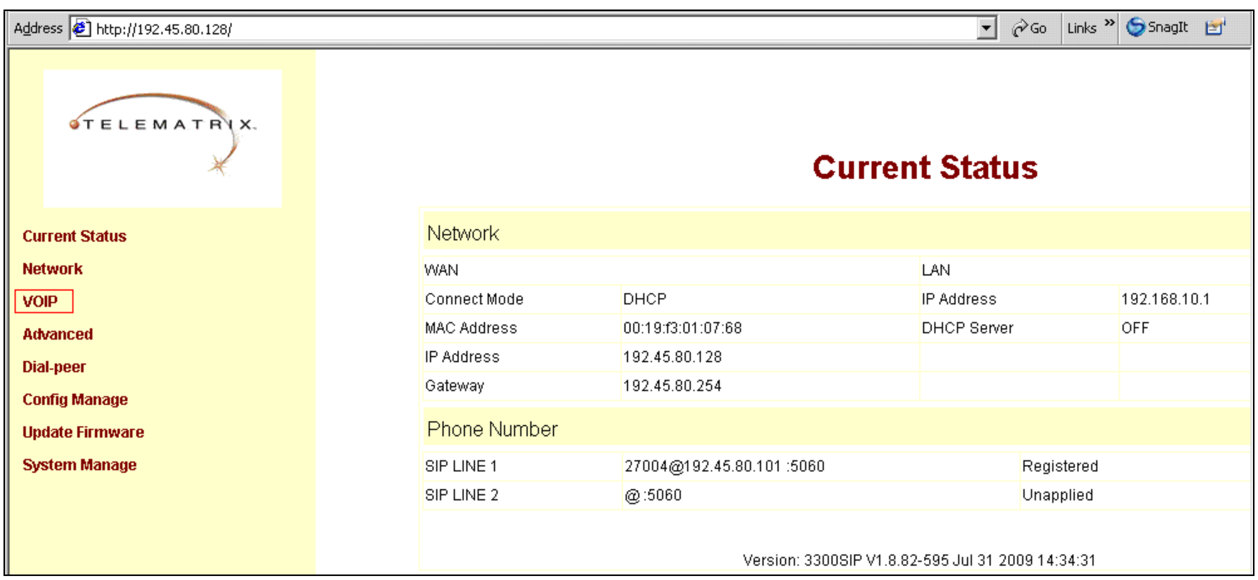


From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's Communication Manager Server from the drop-down list. Click on the **Add** button.



6. Configure Telematrix 3302IP and 9602IP SIP telephones

This section provides steps to configure Telematrix 3302IP and 9602IP SIP telephones. The latest firmware was provided by Telematrix. Once the Telematrix 3302IP and 9602IP SIP telephones have acquired IP addresses from the DHCP server, retrieve the phone's IP address by pressing * * 4(I) 7(P) #. The phone's IP address will be spoken by audio through the speaker and/or displayed on the LCD screen. Launch a web browser, enter **http://<IP address of the Telematrix SIP telephone>** in the URL, and log in with the appropriate credentials (Default login = **admin** / password = **admin**) to access the **Current Status** page. Select the **VOIP → SIP Config** link from the left pane of the screen.

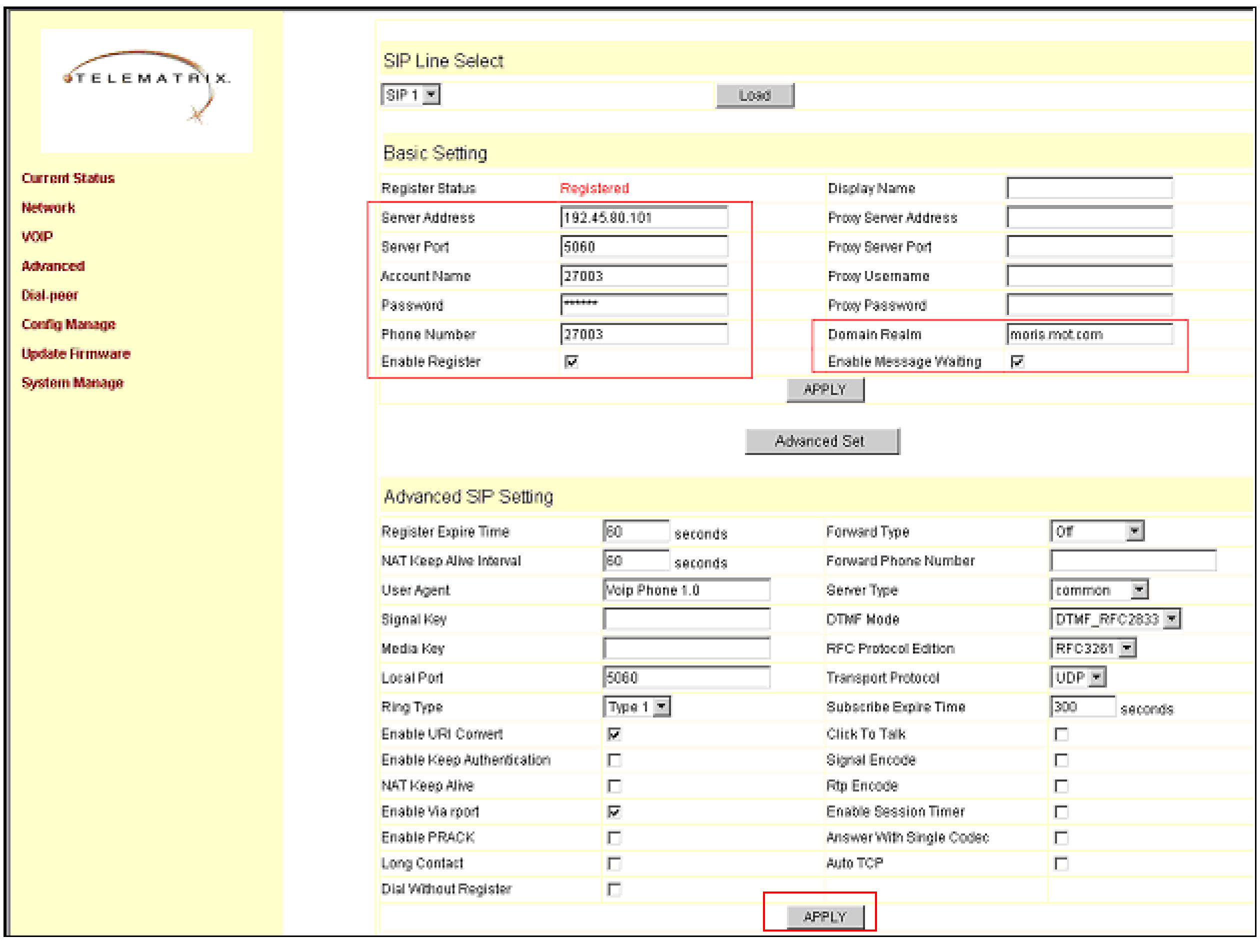


Provide the following information:

- Server Address – Enter the Avaya Aura SES IP address.
- Server Port – Enter the port number used. The default port number is 5060.
- Account Name – Enter the user created in **Section 5.3**.
- Password – Enter the password created for the user.
- Phone Number – Enter the phone number associated with the user.
- Check the **Enable Register** check box.
- Domain Realm – Enter the SIP domain created in **Section 5.1**.
- Check the **Enable Message Waiting** check box.

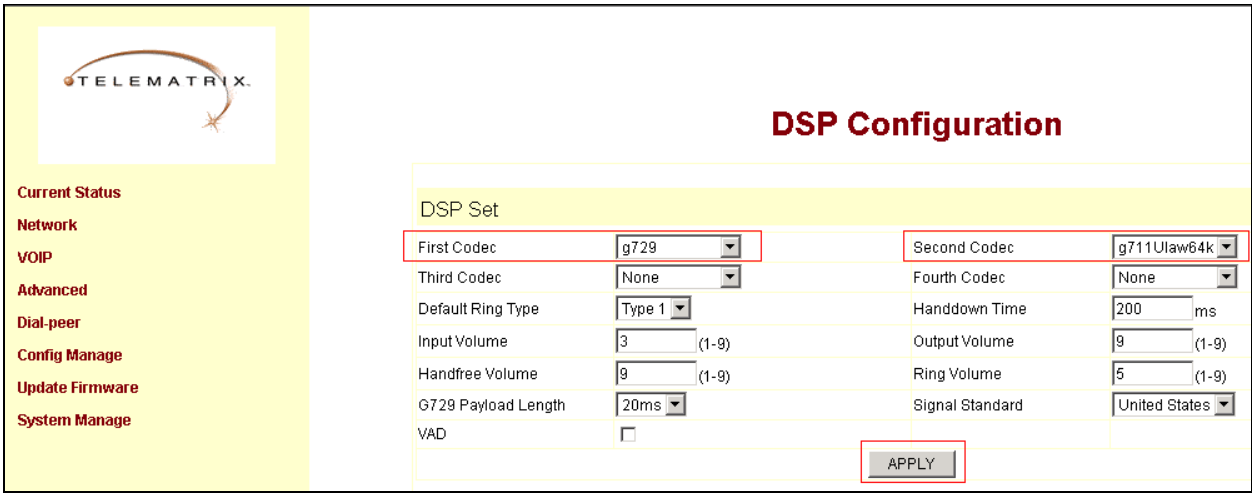
As the Advanced Set button is clicked, the bottom half of the screen will appear. During the compliance test, default values were used.

Click the **APPLY** button to submit changes.



Select the **Advanced → DSP Configuration** link from the left pane of the screen to access the DSP Configuration page. Select the codecs that will be utilized. During the compliance test, G.711Mu and G.729 were utilized. Default values were used for the remaining fields.

Click the **APPLY** button to submit changes.



7. Interoperability Compliance Testing

The interoperability compliance test included verification of features and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on Telematrix 3302IP and 9602IP SIP telephones. Telematrix 3302IP and 9602IP SIP telephones operation such as inbound calls, outbound calls, hold, forward, conference, Feature Name Extension (FNE), and Telematrix 3302IP and 9602IP SIP telephones interaction with Avaya SIP Enablement Services (SES), Avaya Aura Communication Manager, and Avaya SIP, H.323, and digital telephones were verified. The serviceability testing introduced failure scenarios to see if Telematrix 3302IP and 9602IP SIP telephones can recover from failures.

7.1. General Test Approach

The general test approach was to place calls to and from Telematrix 3302IP and 9602IP SIP telephones and exercise basic telephone operations. The main objectives were to verify that:

- Telematrix 3302IP and 9602IP SIP telephones successfully register with Avaya Aura SES.
- Successfully establish calls between Telematrix 3302IP and 9602IP SIP telephones and Avaya SIP, H.323, and digital telephones attached to Avaya Aura SES or Avaya Aura Communication Manager.
- Telematrix 3302IP and 9602IP SIP telephones successfully negotiate the right codec (G.711MU and G.729).
- Telematrix 3302IP and 9602IP SIP telephones successfully hold a call.
- Telematrix 3302IP and 9602IP SIP telephones successfully establish a three party conference call.

- Telematrix 3302IP and 9602IP SIP telephones successfully verify following FNE features:
    - Call Park
    - Call Pickup
    - Auto Redial
    - Last Number Dialed
    - Send All Calls
    - Call Forward (Unconditional, Busy/no answer)
    - Find Me

For serviceability testing, failures such as cable pulls and hardware resets were applied.

7.2. Test Results

The test objectives of **Section 7.1** were verified. For serviceability testing, the Telematrix 3302IP and 9602IP SIP telephones operated properly after recovering from failures such as cable disconnects, and resets of the Telematrix 3302IP and 9602IP SIP telephones and the Avaya Aura SES server. Telematrix 3302IP and 9602IP SIP telephones successfully negotiated the codec to be used.

8. Verification Steps

The following steps may be used to verify the configuration:

The following steps may be used to verify the configuration:

- Verify that Telematrix 3302IP and 9602IP SIP telephones successfully register with Avaya Aura SES server by following the **Users → Registered Users** link on the SES Administration Web Interface.
- Place calls to and from Telematrix 3302IP and 9602IP SIP telephones and verify that the calls are successfully established with two-way talk path.
- While calls are established, Enter **status trunk <t:r>** command, where **t** is the SIP trunk group configured in **Section 4.6**, and **r** is trunk group member. This will verify whether the call is shuffled or not.

9. Conclusion

Telematrix 3302IP and 9602IP SIP telephones were compliance tested with the Avaya Aura Communication Manager (Version 5.2) and Avaya Aura SES (Version 5.2). Telematrix 3302IP and 9602IP SIP telephones functioned properly for feature and serviceability. During compliance testing, Telematrix 3302IP and 9602IP SIP telephones successfully registered with Avaya Aura SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like three-way conference, hold, etc.

Between Telematrix 3302IP and 9602IP SIP telephones and Avaya 9600 Series SIP telephones, the following codecs were verified for operation:

- G.711Mu
- G.729

10. Additional References

The following Avaya product documentation can be found at http://support.avaya.com

[1] *Administering Avaya Aura™ Communication Manager* Release 5.2, Issue 5, May 2009, Document Number 03-300509.
[2] *Avaya Aura™ Communication Manager Screen Reference,* Issue 1.0, May 2009, Document Number 03-602878.
[3] *Administering Avaya Aura™ SIP Enablement Services on the Avaya S8300 Server,* Issue 2.0, May 2008, Document Number 03-602508.

The following document was provided by Telematrix.

[4] *3300IP VOIP Phone User Manual*.
[5] *Telematrix SIP Phone Administration Guide*
[6] *Telematrix SIP QuickStart Guide*

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 DevConnect Program at devconnect@avaya.com.