Avaya Solution & Interoperability Test Lab

Application Notes for Tenacity ipTTY and Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Tenacity ipTTY, which was compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services.

The overall objective of the interoperability compliance testing is to verify Tenacity ipTTY functionalities in an environment comprised of Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, and various 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 Tenacity ipTTY, which was compliance tested with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services (SES).

Tenacity ipTTY is engineered to enable TTY communications using an existing VoIP/Hybrid PBX infrastructure. The only requirement from the infrastructure is support for 3rd party SIP devices. With Tenacity ipTTY, there is no longer a need for outdated TTY machines or expensive computer modems. Most importantly, with ipTTY, analog telephone lines are not required to facilitate TTY communications. Additionally, the ipTTY supports Hearing Carry Over (HCO), Voice Carry Over (VCO), includes a multi-lined display (versus a single lined display like standard TTY machines) and offers a recent calls list.

These Application Notes assume that Communication Manager and SES are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. In these Application Notes, the following topics will be described:

- Communication Manager – SIP trunk configuration
- SIP Enablement Services – User configuration
- Tenacity – User registration

For further details on configuration steps not covered in this document, consult [3].

1.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on the Tenacity ipTTY. Tenacity ipTTY operations such as inbound calls, outbound calls, call transfer, call forward, DTMF, and Tenacity ipTTY interactions with SES, Communication Manager, and Avaya SIP and H.323 telephones were verified. The serviceability testing introduced failure scenarios to see if Tenacity ipTTY can recover from failures.

1.2. Support

Technical support for Tenacity ipTTY solution can be obtained by contacting Tenacity: support@accessaphone.com or (866)756-0321.

2. Reference Configuration

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G450 Media Gateway, an SES server, and Tenacity ipTTY. The solution described herein is also extensible to other Avaya Servers and Media Gateways. Avaya S8720 Servers with an Avaya G650 Media Gateway were included in the test to provide an inter-switch scenario. For completeness, Avaya 4600 Series H.323 IP Telephones, Avaya 9600 Series SIP IP Telephones, and Avaya 9600 Series H.323 IP Telephones are included in **Figure 1** to demonstrate calls between the SIP-based Tenacity ipTTY and Avaya SIP, H.323, and digital telephones.

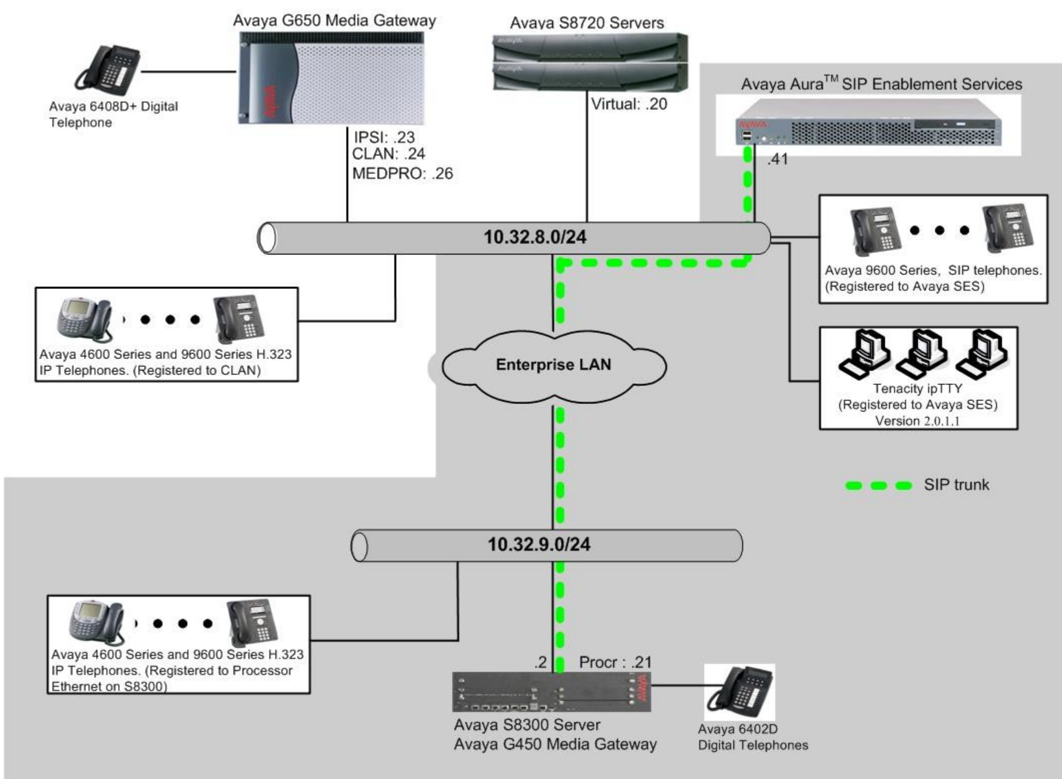


Figure 1: Test Configuration of Tenacity ipTTY

3. Equipment and Software Validated

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

Equipment		Software/Firmware
Avaya S8720 Servers		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya G650 Media Gateway		-
	TN2312BP IP Server Interface TN799DP C-LAN Interface TN2302AP IP Media Processor	HW11 FW044 HW01 FW028 HW20 FW118
Avaya S8300 Server with Avaya G450 Media Gateway		Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3)
Avaya Aura™ SIP Enablement Services on S8500 Server		Avaya Aura™ SIP Enablement Services 5.2 (R015x.02.0.947.3) with Service Pack SES-02.0.947.3-SP2a
Avaya 4600 and 9600 Series SIP Telephones		
	9620 (SIP) 9630 (SIP) 9650 (SIP)	2.0.5 2.0.5 2.0.5
Avaya 4600 and 9600 Series IP Telephones		
	4625 (H.323) 9630 (H.323) 9650 (H.323)	2.9 3.002 3.002
Avaya 6408D+ Digital Telephone		-
Tenacity ipTTY		2.0.1.1

4. Configure Avaya Aura™ Communication Manager

This section describes the procedure for setting up a SIP trunk between Communication Manager and SES for the support of SIP stations. The steps include setting up an IP codec set, an IP network region, 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 Communication Manager System Access Terminal (SAT) interface. Tenacity ipTTY and other SIP telephones are configured as off-PBX telephones in 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   72
                                      Maximum Stations: 450   19
                              Maximum XMOBILE Stations: 45    0
                     Maximum Off-PBX Telephones - EC500: 50    0
                     Maximum Off-PBX Telephones -   OPS: 100   10
                     Maximum Off-PBX Telephones - PBFMC: 45    0
                     Maximum Off-PBX Telephones - PVFMC: 45    0
                     Maximum Off-PBX Telephones - SCCAN: 45    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   20
           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: 100   0
                  Maximum Video Capable H.323 Stations: 5     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 100   20
  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 Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and 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 the IP network region.

Since Tenacity ipTTY only supports G.711MU, G.711MU was the only codec included in the ip-codec-set form.

```
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:
```

4.3. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and SES. In the sample configuration, for simplicity, a single network region was used. See the note below the screen for additional considerations.

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. During the compliance test, the authoritative domain is set to **testroom.avaya.com**. This should match the SIP Domain value on SES, in **Section 5.1**.
- **Intra-region IP-IP Direct Audio** – Set to **no** to deny direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or 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 **no** to deny direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or 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: testroom.avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   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
```

In production environments, multiple network regions can be configured so that intra-region and inter-region behaviors can be distinguished and customized. For example, the codec set used for intra-region communications can differ from the codec set used for communications between specific inter-region pairs. Also, since shuffling to IP-IP Direct Audio is disabled to accommodate Tenacity ipTTY, having multiple network regions can allow Avaya devices to continue to shuffle to IP-IP Direct Audio, while still preventing shuffling for connections involving Tenacity ipTTY.

4.4. Configure IP Node Name

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

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
CLAN                    10.32.8.24
IA770                   10.32.9.22
SES                     10.32.8.41
default                 0.0.0.0
procr                   10.32.9.21
```

4.5. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between 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.**
- **Transport Method** – Set to **tls**
- **Near-end Node Name** - Set to **procr** as displayed in **Section 4.4**.
- **Far-end Node Name** - Set to the 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 **testroom.avaya.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
IMS Enabled? n
   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: testroom.avaya.com
                                             Bypass If IP Threshold Exceeded? n

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

4.6. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and 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 –** 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 Communication Manager and associating the OPS station extensions with the telephone numbers of Tenacity ipTTY. 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 for Tenacity ipTTY.

```
add station 27003                                               Page   1 of   6
                                     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: 19       Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 27003
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n



                                               Customizable Labels? y
```

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 Tenacity ipTTY 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
```

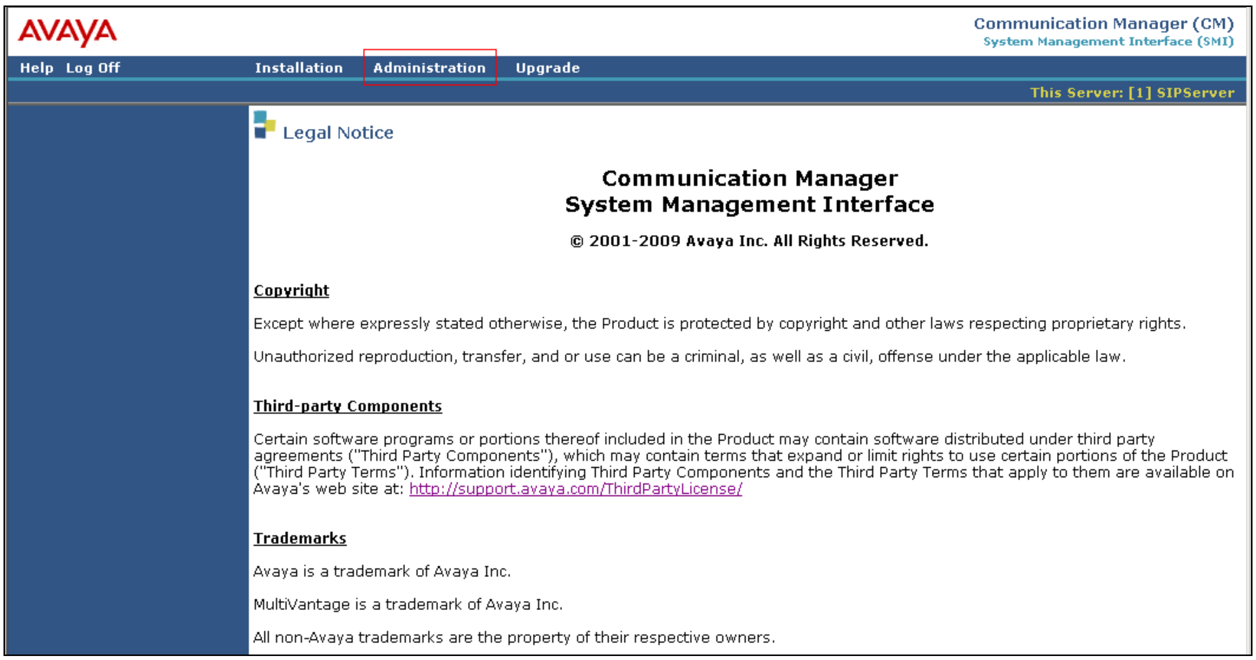
5. Configure Avaya Aura^{TM} SIP Enablement Services

This section describes the steps for creating SIP trunks between SES and Communication Manager. SIP user accounts are configured in SES and associated with Communication Manager OPS station extensions. During the compliance test, Tenacity ipTTY is treated as a SIP endpoint. The highlights in the following screens indicate the values used during the compliance test. Default values may be used for all other fields.

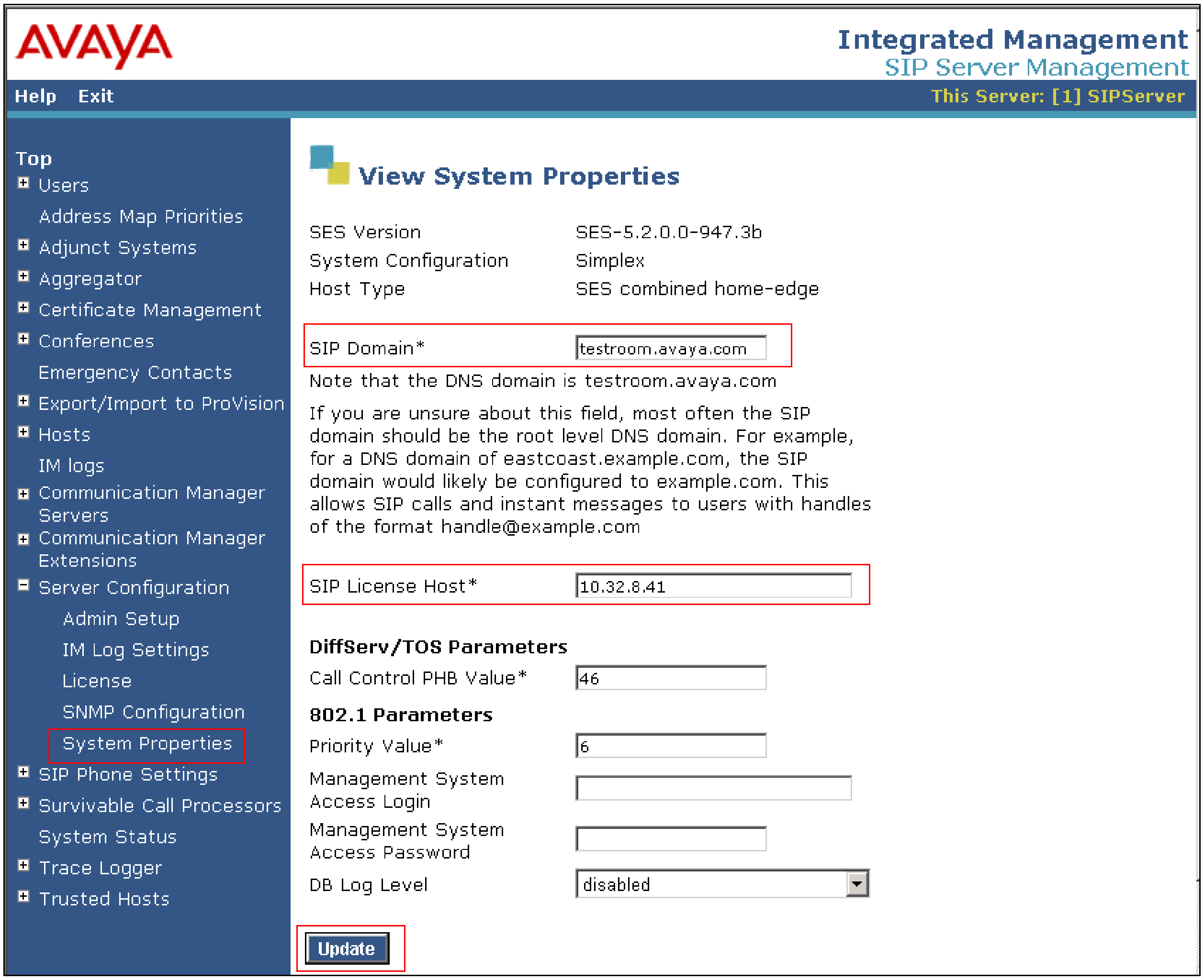
5.1. Configure SES Server Properties

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 Communication Manager in **Section 4.5**. Click on the **Update** button, after the completion.

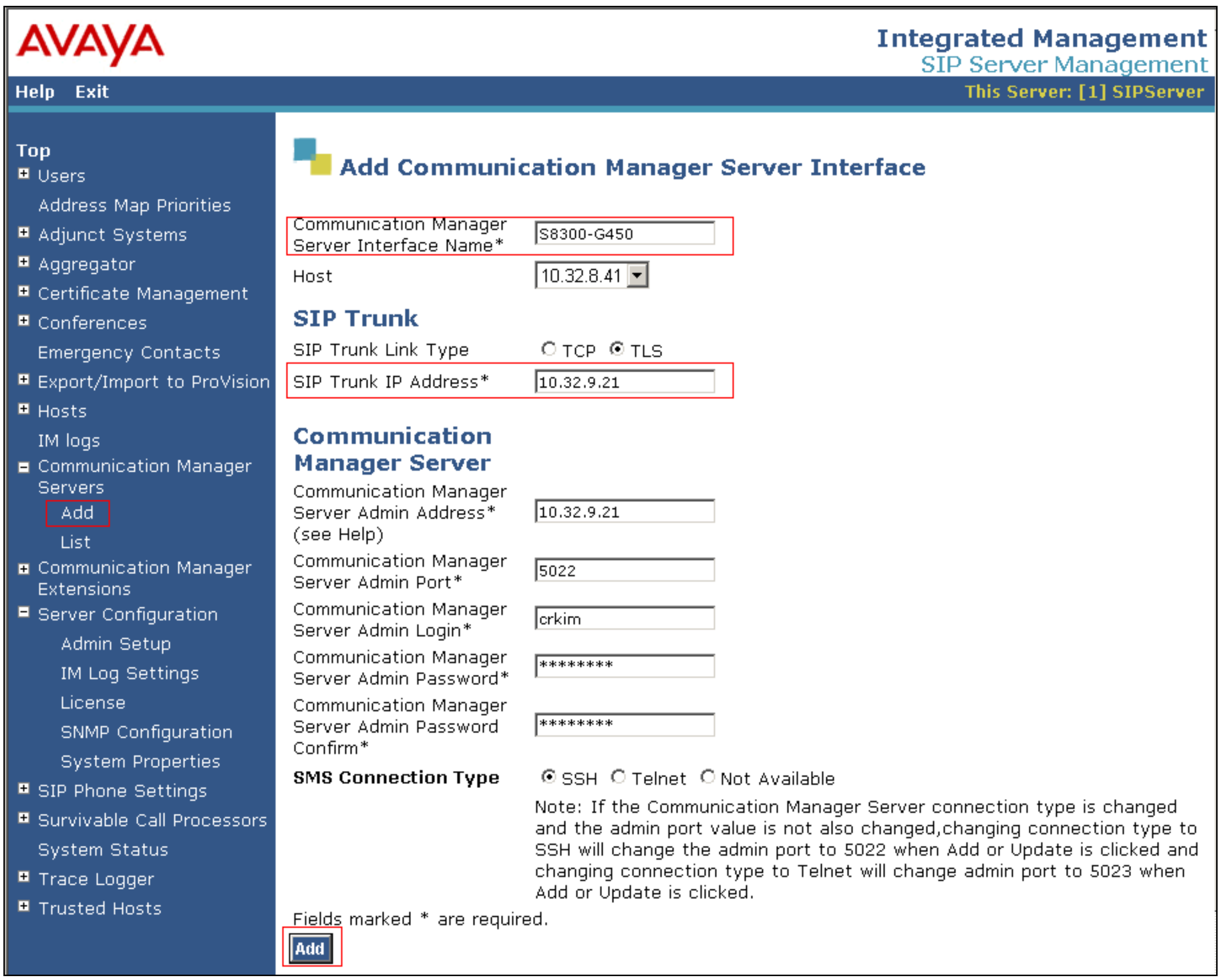


5.2. Configure Communication Manager Servers

This section provides steps to add SIP-enabled 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 **Add Communication Manager 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 Processor Ethernet (procr) IP interface that terminates the SIP link from SES.

Click **Add** when finished.

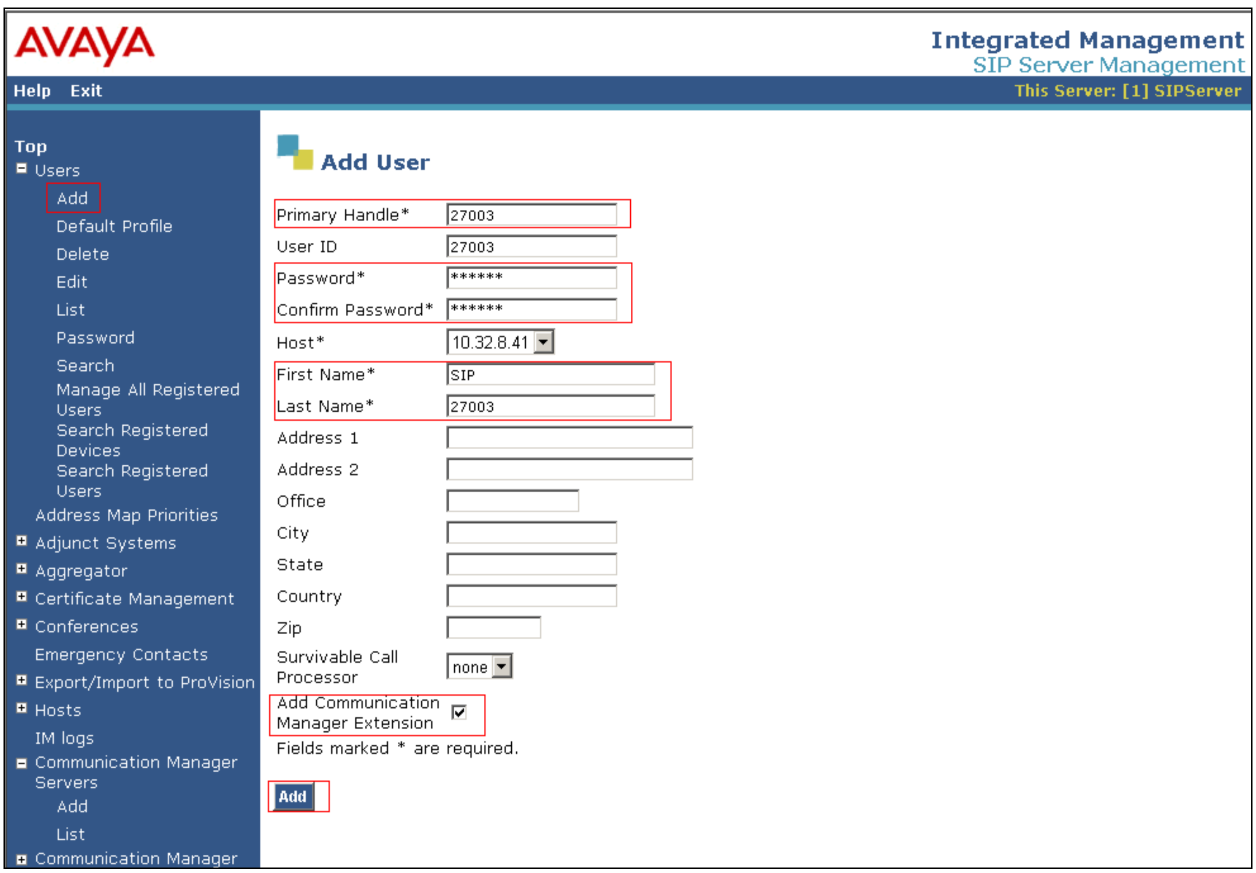


5.3. Configure Users

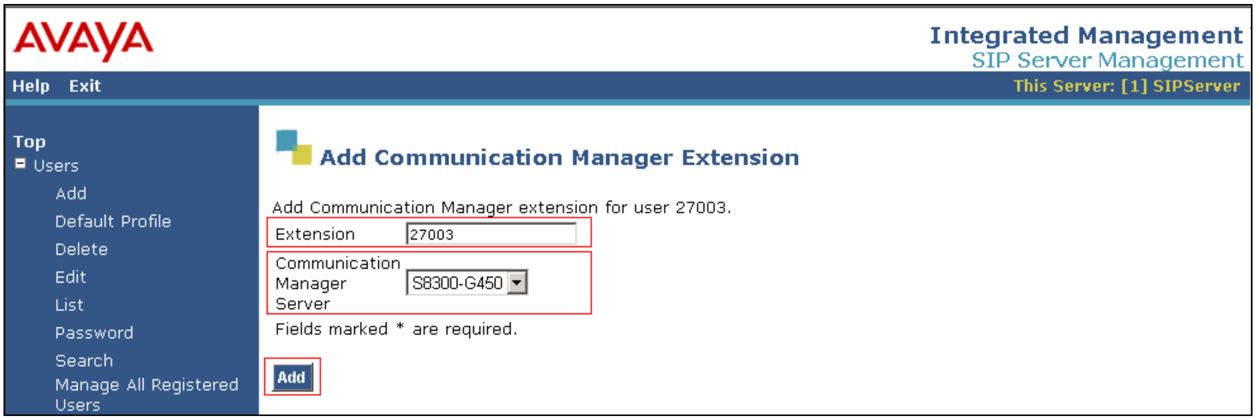
This section provides steps to add users to be administered in the SES database. In the **Integrated Management SIP Server Management** page, select the **Users → Add** link from the left pane of the screen. The highlighted fields were configured for the compliance test:

- **Primary Handle** – Enter the phone number to be used for a Tenacity ipTTY user. This number was configured in **Section 4.7**.
- **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 Communication Manager 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, as shown in the subsequent screen. If not, an extension can be associated with the user in the future.

Click **Add** when finished.

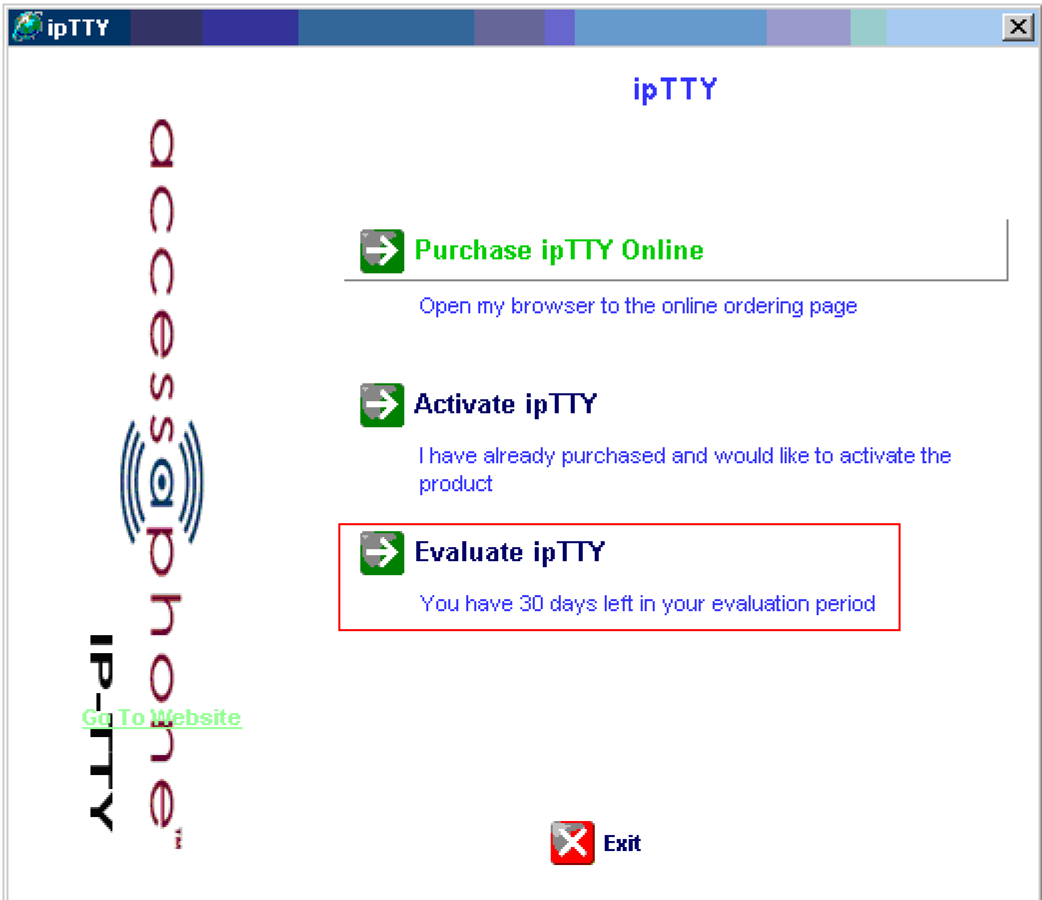


From the next screen, enter the numeric telephone extension you want to create in the database. Select the extension's Communication Manager Server from the drop-down list. Click on the **Add** button.

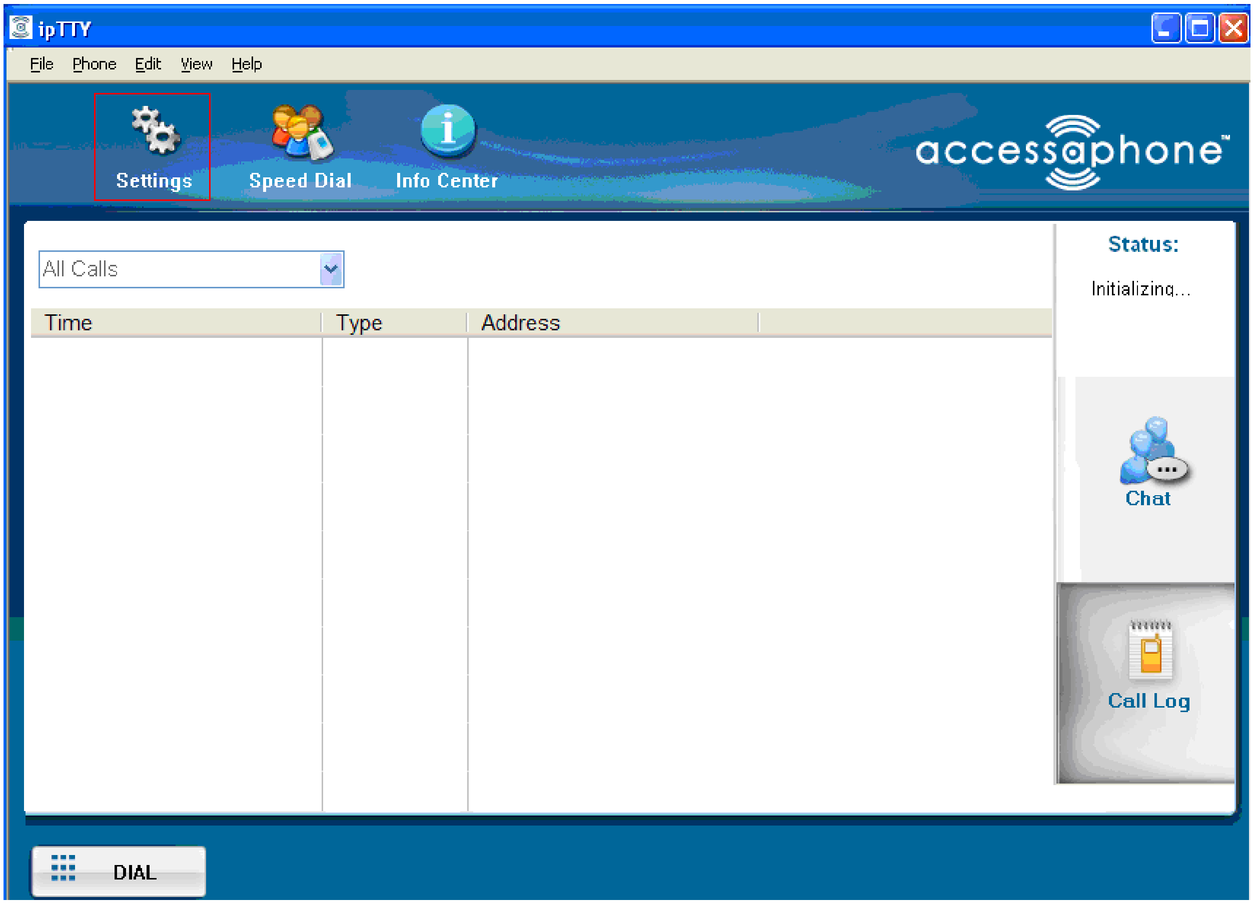


6. Configure Tenacity ipTTY

This section provides steps to configure Tenacity ipTTY. The latest firmware was provided by Tenacity. To start Tenacity ipTTY application, double click the ipTTY icon ( ). During the compliance test, the **Evaluate ipTTY** license was utilized.



Select the **Settings** button to configure the ipTTY for interfacing with SES.

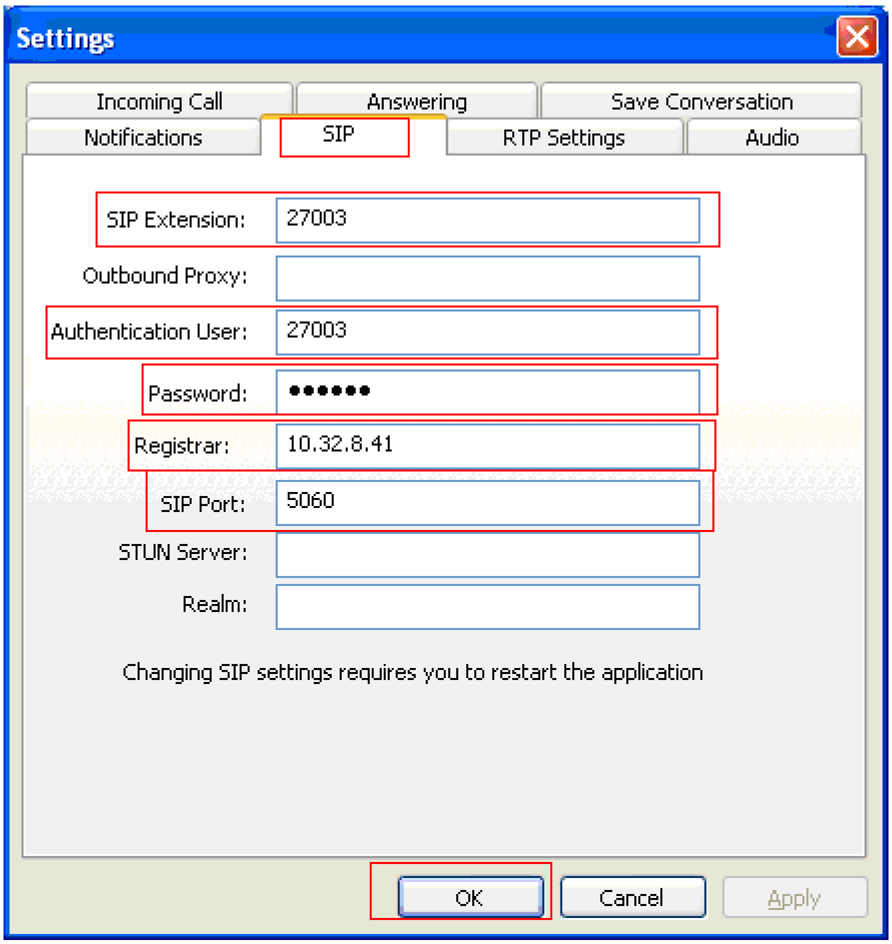


Select the **SIP** tab, and provide the following information:

- SIP Extension – Enter the user extension created in **Section 5.3**.
- Authentication User – username (usually the same as the SIP Extension)
- Password – Enter the password created in **Section 5.3**.
- Registrar – Enter the IP address of SES.
- SIP Port – The default port is utilized.

Click on the **OK** button, after the completion.

Note*: In order for the settings to take effect, the ipTTY must be closed and reopened.*



7. General Test Approach and Test Results

The general test approach was to place calls to and from Tenacity ipTTY and exercise basic telephone operations. The main objectives were to verify that:

- Tenacity ipTTY successfully registers with SES.
- Calls can be successfully established between Tenacity ipTTY and Avaya SIP and H.323 telephones.

- Tenacity ipTTY successfully negotiates the right codec (G.711MU). Tenacity ipTTY supports only G.711MU codec.
- Tenacity ipTTY successfully transfers a call (Origination).
- Tenacity ipTTY successfully forwards a call.

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

The test objectives were verified. For serviceability testing, the Tenacity ipTTY operated properly after recovering from failures such as cable disconnects, and resets of the Tenacity ipTTY and the SES server. To make Tenacity ipTTY interoperable in an Avaya VoIP environment, shuffling for connections involving ipTTY must be disabled.

8. Verification Steps

The following steps may be used to verify the configuration:

- Verify that Tenacity ipTTY successfully registers with SES server by following the **Users → Registered Users** link on the SES Administration Web Interface.
- Place calls to and from Tenacity ipTTY 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 the trunk group member used for a call. This can verify whether the call is shuffled or not.

9. Conclusion

Tenacity ipTTY was compliance tested with Communication Manager (Version 5.2) and SES (Version 5.2). Tenacity ipTTY (Version 2.0.1.1) functioned properly for feature and serviceability. During compliance testing, Tenacity ipTTY successfully registered with SES, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features like source transfer.

Note: *During the compliance test, the source transfer function was successful. However, the destination transfer was not successful. This issue is being investigated.*

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] SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers, Release 5.2, Issue 9, May 2009, Document Number 555-245-206

The following document was provided by Tenacity.

[3] *Tenacity ipTTY Quick Start Guide*, Document Version 1.2, October 2009.

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.