



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Flying Voice Technology IP652 SIP Phones with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Flying Voice Technology IP652 SIP Phones to interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1.

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 steps required to configure Flying Voice Technology IP652 SIP Phones to interoperate with an Avaya SIP infrastructure consisting of Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. The IP652 is 5-line business IP phone for the enterprise.

2. General Test Approach and Test Results

To verify interoperability of Flying Voice IP652 SIP Phones with Session Manager and Communication Manager, calls were made between Flying Voice telephones and Avaya SIP, H.323 and Digital telephones using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using speed-dial buttons.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of Flying Voice telephones with Session Manager.
- Calls between Flying Voice telephones and Avaya SIP, H.323, and digital telephones.
- G.722, G.711 and G.729A codec support and negotiation, with and without media shuffling.
- Basic features including phone display, mute/un-mute, answer, hang up, music on hold, DTMF transmission, Message Waiting Indicator (MWI) subscription and feature access code dialing.
- PBX features including Multiple Call Appearances, Hold, Transfer, and Conference.
- Proper system recovery after a Flying Voice telephone restart and loss of IP connection.

2.2. Test Results

All test cases were executed. The following were observations on Flying Voice IP652 from the compliance testing.

- For G.729A codec, calls with Avaya SIP users did not shuffle.
- For G.729A codec, calls between IP652 phones did not shuffle.
- For outgoing calls, DTMF was sent in-band as the RFC2833 negotiation was not successful.

2.3. Support

Technical support from Flying Voice Technology can be obtained through the following:

- Phone: +86-755-26099365
- E-mail: support@e3call.com

3. Reference Configuration

The diagram illustrates an enterprise site with an Avaya SIP-based network, including Session Manager, an S8800 Server running Communication Manager with a G450 Media Gateway, and Avaya SIP, H.323 and Digital endpoints. The enterprise site also contains two Flying Voice IP652 SIP Phones used in the compliance testing. The Flying Voice phones are registered with Session Manager and are configured as endpoint users.

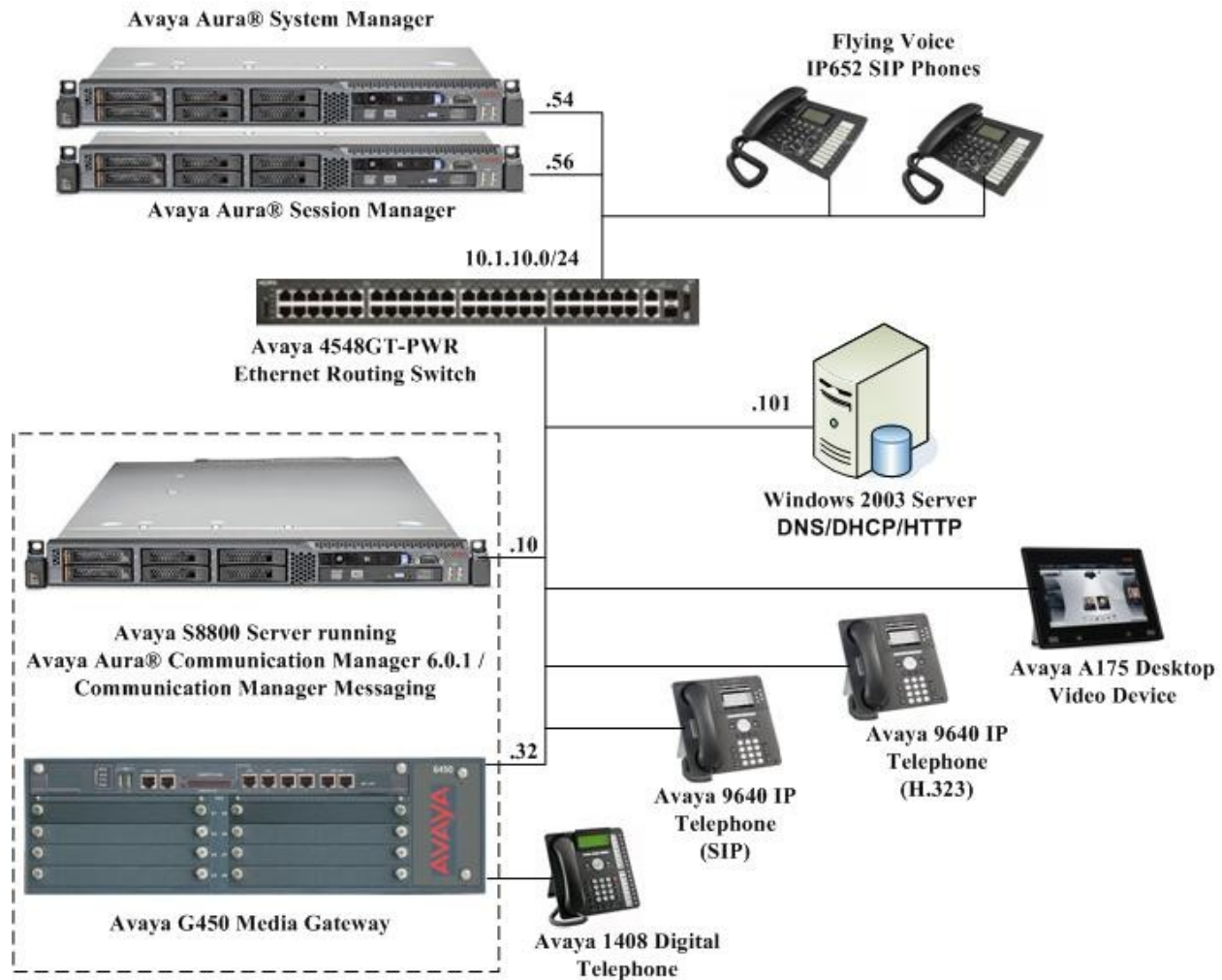


Figure 1: Flying Voice IP652 Phones with Avaya SIP Solution

Table 1 lists the extensions used for this testing.

| Extension | Note |
|------------------|---|
| 10099 | Avaya 9640 IP Telephone (H.323) |
| 10051 | Avaya 9640 IP Telephone (SIP) |
| 10016 | Avaya 1408 Digital Telephone |
| 10061 | Avaya Desktop Video Device (ADVD) (SIP) |
| 10067 to 10068 | Flying Voice IP652 SIP Phones |

Table 1 – Extension Setup

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

| Equipment | Software |
|---|--|
| Avaya S8800 Server | Avaya Aura® Communication Manager 6.0.1 (Service Pack 4 00.1.510.1-19100) / Avaya Aura® Communication Manager Messaging 6.0.1 |
| Avaya G450 Media Gateway | 31.20.0 |
| Avaya S8800 Server | Avaya Aura® Session Manager 6.1 Service Pack 2 |
| Avaya S8800 Server | Avaya Aura® System Manager 6.1 Service Pack 2 |
| Avaya 9600 Series IP Telephones | 2.6.4.0 (SIP) 3.1 SP2 (H.323) |
| Avaya 1408 Digital Telephone | - |
| Avaya Desktop Video Device | 1.0.3 |
| Avaya 4548GT-PWR Ethernet Routing Switch | V5.4.0.008 |
| Flying Voice IP652 SIP Phones | Hardware Version: 1.0.1 Firmware Version: 1.3.4 (Hy0811225428) DSP Version: D2.63 |

5. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer users

5.1. Launch Avaya Aura® System Manager

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

The screenshot shows the Avaya Aura® System Manager 6.1 login interface. At the top, the Avaya logo is on the left and the title 'Avaya Aura® System Manager 6.1' is on the right. Below the title bar is a red navigation bar with 'Home / Log On'. The main heading is 'Log On'. On the left, a box contains instructions: 'Recommended access to System Manager is via FQDN.' followed by a link 'Go to central login for Single Sign-On'. Below this, it states 'If IP address access is your only option, then note that authentication will fail in the following cases:' and lists two bullet points: 'First time login with "admin" account' and 'Expired/Reset passwords'. On the right, there are input fields for 'User ID:' and 'Password:'. At the bottom right are 'Log On' and 'Cancel' buttons, and a 'Change Password' link at the very bottom right.

 Avaya Aura® System Manager 6.1

Home / Log On

Log On

Recommended access to System Manager is via FQDN.

[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

User ID:

Password:

Log On Cancel

[Change Password](#)

5.2. Administer Users

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.

The screenshot shows the Avaya Aura System Manager 6.1 User Management interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.1', and links for 'Help', 'About', 'Change Password', and 'Log off admin'. Below this is a breadcrumb trail: 'Home / Users / User Management / Manage Users - User Management'. The left sidebar contains a 'User Management' dropdown menu with options: 'Manage Users' (highlighted), 'Public Contacts', 'Shared Addresses', and 'System Presence ACLs'. The main content area is titled 'User Management' and features a 'Users' section with buttons for 'View', 'Edit', 'New' (circled in red), 'Duplicate', 'Delete', and 'More Actions'. Below these buttons is a table with 22 items, showing a list of users. The table has columns for 'Status', 'Name', 'Login Name', 'E164 Handle', and 'Last Login'. The 'New' button is circled in red, indicating the action to be taken.

| Status | Name | Login Name | E164 Handle | Last Login |
|--------------|-----------------|------------|-------------|------------|
| 1XC SIPUser1 | 10063@sglab.com | 10063 | | |
| 1XC SIPUser2 | 10064@sglab.com | 10064 | | |
| ADVD User1 | 10061@sglab.com | 10061 | | |
| ADVD User2 | 10062@sglab.com | 10062 | | |
| Avaya, SIP1 | 10051@sglab.com | 10051 | | |

5.2.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter **n@x**, where **n** is the first IP652 user extension and **x** is the domain name used for compliance testing, in this case **sglab.com**. For **Password** and **Confirm Password**, enter the appropriate credentials for System Manager. Retain the default values in the remaining fields.

The screenshot displays the Avaya Aura System Manager 6.1 interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off admin". A secondary navigation bar shows "User Management" and "Home". The left sidebar contains a "User Management" dropdown menu with options: "Manage Users", "Public Contacts", "Shared Addresses", and "System Presence ACLs". The main content area is titled "New User Profile" and features a breadcrumb trail: "Home / Users / User Management / Manage Users - New User Profile". Below the title are "Commit" and "Cancel" buttons. The "New User Profile" section has four tabs: "Identity", "Communication Profile", "Membership", and "Contacts". The "Identity" tab is selected, showing the following fields:

- * Last Name: Doe
- * First Name: John
- Middle Name: (empty)
- Description: (empty)
- * Login Name: 10067@sglab.com
- * Authentication Type: Basic
- * Password: (masked with dots)
- * Confirm Password: (masked with dots)

5.2.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

User Management * Home

Home /Users / User Management / Manage Users- New User Profile

Help ?

New User Profile [Commit] [Cancel]

Identity * Communication Profile * Membership Contacts

Communication Profile

Communication Profile Password:

Confirm Password:

[New] [Delete] [Done] [Cancel]

Name

☒ Primary

Select : None

* Name: Primary

Default : ☒

Communication Address

[New] [Edit] [Delete]

| Type | Handle | Domain |
|------------------|--------|--------|
| No Records found | | |

For **Type**, retain **Avaya SIP**. For **Fully Qualified Address**, enter and select the SIP user extension and domain name from **Section 5.2.1**. Click **Add**.

Communication Address

[New] [Edit] [Delete]

| Type | Handle | Domain |
|------------------|--------|--------|
| No Records found | | |

Type: Avaya SIP

* Fully Qualified Address: 10067 @ sglab.com

[Add] [Cancel]

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Scroll down to check and expand **Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 5.2.1**. For **Template**, select **DEFAULT_9630SIP_CM_6_0**. For **Port**, select **IP**. Retain the default values in the remaining fields.

☒ **Session Manager Profile**

*** Primary Session Manager** me1-sm

| Primary | Secondary | Maximum |
|---------|-----------|---------|
| 22 | 0 | 22 |

Secondary Session Manager (None)

| Primary | Secondary | Maximum |
|---------|-----------|---------|
| | | |

Origination Application Sequence me1-cm-app-seq

Termination Application Sequence me1-cm-app-seq

Survivability Server (None)

*** Home Location** Location1

☒ **Endpoint Profile**

*** System** me1-cm

*** Profile Type** Endpoint

Use Existing Endpoints ☐

*** Extension** 10067 Endpoint Editor

*** Template** DEFAULT_9630SIP_CM_6_0

Set Type 9630SIP

Security Code

*** Port** IP

Voice Mail Number

Delete Endpoint on Unassign of Endpoint from User or on Delete User. ☐

Click **Commit** to complete the creation of the new user (not shown).

Repeat **Section 5.2** to add a user for each Flying Voice IP652 user. In the compliance testing, two users with extensions **10067** and **10068** were added.

6. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Administer IP codec set

6.1. Verify Avaya Aura® Communication Manager License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command to verify that there is sufficient capacity for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the number of Flying Voice IP652 SIP Phone extensions.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

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

G3 Version: V16                                     Software Package: Enterprise
Location: 2                                           System ID (SID): 1
Platform: 28                                          Module ID (MID): 1

                                USED
Platform Maximum Ports: 6400 281
Maximum Stations: 1000 173
Maximum XMOBILE Stations: 2400 0
Maximum Off-PBX Telephones - EC500: 250 0
Maximum Off-PBX Telephones - OPS: 1000 32
Maximum Off-PBX Telephones - PBFMC: 250 0
Maximum Off-PBX Telephones - PVFMC: 250 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 10 1

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

6.2. Administer IP Codec Set

Use the **change ip-codec-set n** command, where **n** is the existing codec set number associated with the SIP trunk group to Session Manager. Update the audio codec types in the **Audio Codec** fields as necessary to include G.722-64K, G.711A, G.711MU and G.729AB.

| | | | | | | |
|-----------------------|-----------|-------------|---------|-----------|------|---|
| 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.722-64K | | 2 | 20 | | |
| 2: | G.711MU | n | 2 | 20 | | |
| 3: | G.711A | n | 2 | 20 | | |
| 4: | G.729AB | n | 2 | 20 | | |

7. Configure Flying Voice IP652 SIP Phones

This section provides the procedures for configuring Flying Voice IP652 SIP Phones. The procedures include the following areas:

- Access Web Interface
- Configure SIP Account

7.1. Access Web Interface

Enter <http://<ip-addr>:8080/>, where <ip-addr> is the IP address of the IP652 phone, into the address bar of web browser and log in using a valid account. The **Status** screen is displayed.

The screenshot displays the 'Status' page of the VOIP-IP652 control panel. The page has a dark header with the 'VOIP-IP652' logo and 'control panel' text. On the right, it shows 'Firmware Version: 1.3.4 (Hy0811225428)', 'DSP Version: D2.63', 'Current Time: Oct 28 13:45:47 2011', and 'Admin Mode [Logout]'. Below the header is a navigation bar with tabs: 'Status' (selected), 'SIP Account', 'Network', 'Phone', and 'Administration'. Under 'Status', there are sub-tabs: 'Basic' (selected), 'DHCP', and 'Syslog'. The main content area is divided into three sections: 'Product Information', 'Line Status', and 'Network Status'. The 'Product Information' section lists details like Product Name (IP652), MAC addresses, and versions. The 'Line Status' section shows the status of five lines (Line 1 is Online, others are Disabled). The 'Network Status' section shows connection details for the Internet port. On the right side, there is a 'Help' section with descriptions for 'Product Information:', 'Line Status:', 'Network Status:', and 'System Status:'.

| Product Information | |
|----------------------------|----------------------|
| Product Name: | IP652 |
| Internet(WAN) MAC Address: | 00:21:F2:04:2E:27 |
| PC(LAN) MAC Address: | 00:21:F2:04:2E:26 |
| Hardware Version: | 1.0.1 |
| Firmware Version: | 1.3.4 (Hy0811225428) |
| DSP Version: | D2.63 |

| Line Status | |
|----------------|---------|
| Line 1 Status: | Online |
| Line 2 Status: | Disable |
| Line 3 Status: | Disable |
| Line 4 Status: | Disable |
| Line 5 Status: | Disable |

| Network Status | |
|----------------------|----------------|
| Internet Port Status | |
| Connection Status: | Connected |
| Connection Type: | DHCP |
| IP Address: | 10.1.10.178 |
| Subnet Mask: | 255.255.255.0 |
| Default Gateway: | 10.1.10.1 |
| Primary DNS: | 10.1.10.101 |
| Secondary DNS: | 219.141.140.10 |

Help

Product Information:
It shows the basic information of the product.

Line Status:
It shows the registration state of each line.

Network Status:
It shows the information of WAN port, VPN and LAN port.

System Status:
It shows the current time and the running time of the product.

7.2. Configure SIP Account

Select **SIP Account** → **Line 1** from the top menu. Set the **Line Enable** field to **Enable**. Enter the IP address of Session Manager signaling interface as shown in **Figure 1** for the **Proxy Server** field. For the fields **Account**, **Phone Number** and **Password**, enter the account details as shown below to match the User settings in Session Manager added in **Section 5.2**.

In the **Codec Setup** section, prioritize the audio codecs accordingly.

The screenshot displays the 'VOIP-IP652 ... control panel' interface. At the top, it shows 'Firmware Version: 1.3.4 (Hy0811225428)', 'DSP Version: D2.63', 'Current Time: Oct 28 13:48:18 2011', and 'Admin Mode [Logout]'. The main navigation bar includes 'Status', 'SIP Account', 'Network', 'Phone', and 'Administration'. Below this, a sub-menu shows 'SIP Settings', 'Line 1', 'Line 2', 'Line 3', 'Line 4', and 'Line 5'. The 'Line 1' tab is active, showing the 'Basic' configuration section. This section includes 'Basic Setup' (Line Enable: Enable, Peer To Peer: Disable), 'Proxy and Registration' (Proxy Server: 10.1.10.56, Proxy Port: 5060, Outbound Server: , Outbound Port: 5060, Backup Outbound Server: , Backup Outbound Port: 5060), and 'Subscriber Information' (Display Name: IP652 One, Account: 10067, Phone Number: 10067, Password:). Below the 'Basic' section is the 'Audio Configuration' section, which includes 'Codec Setup' (Audio Codec Type 1: G.711U, Audio Codec Type 2: G.711A, Audio Codec Type 3: G.729, Audio Codec Type 4: G.722, Audio Codec Type 5: G.723, G.723 Coding Speed: 5.3k bps, Packet Cycle(ms): 20ms, Silence Supp: Disable, Echo Cancel: Disable). On the right side, there is a 'Help' section with 'Basic' (Set the basic information provided by your VOIP Service Provider, such as Phone Number, Account, password, SIP Proxy and so on.), 'Audio Configuration' (Select the audio Codec you want to use.), 'Supplementary Service Subscription' (Call Waiting - This call feature allows your phone to accept other incoming calls during the conversation.), and 'Advanced' (The Advanced parameters for Administrator.).

In the **Supplementary Service** section, ensure that the **MWI Enable** field is set to **Enable**.

In the **Advanced Setup** section, set **DTMF Type** to **RFC2833**. The rest of the fields can be left at their default values. Click **Save Settings**, then followed by **Reboot** to restart the phone. This completes the configuration required for Flying Voice IP652 SIP Phones.

| Supplementary Service Subscription | | | |
|--|-----------------------|---------------------------------|----------------------|
| Supplementary Services | | | |
| Call Waiting: | Enable ▾ | | |
| Dial Prefix: | <input type="text"/> | Delayed Hot Line: | <input type="text"/> |
| MWI Enable: | Enable ▾ | Voice Mailbox Numbers: | 10000 |
| Advanced | | | |
| Advanced Setup | | | |
| Domain Name Type: | Enable ▾ | Carry Port Information: | Enable ▾ |
| Signal Port: | 5060 | DTMF Type: | RFC2833 ▾ |
| RFC2833 Payload(>=96): | 101 | Register Refresh Interval(sec): | 3600 |
| RTP Port: | 0 (=0 auto select) | Cancel Message Enable: | Disable ▾ |
| Session Refresh Time(sec): | 180 | Refresher: | UAC ▾ |
| Prack Enable: | Disable ▾ | SIP Ping Enable: | Disable ▾ |
| Keep-alive Interval(10-60s): | 15 | Anonymous Call: | Disable ▾ |
| Anonymous Call Block: | Disable ▾ | Proxy DNS Type: | A Type ▾ |
| Use OB Proxy In Dialog | Disable ▾ | VPN: | Disable ▾ |
| <div>Save Settings Cancel Changes Reboot</div> | | | |

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Session Manager and Flying Voice IP652 SIP Phones.

From the System Manager Web interface, select **Elements → Session Manager → System Status → User Registrations** to display the **User Registrations** screen. Verify that the users from **Section 5.2** are registered, as shown below with a check in the **Registered Prim** column.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager x Home

Home / Elements / Session Manager / System Status / User Registrations - User Registrations

Help ?

User Registrations

Select rows to send notifications to AST devices. Click on Details column for complete registration status.

Customize

AST Device Notifications: Reboot Reload Failback As of 12:59 PM

Advanced Search

21 Items Refresh Show 15 Filter: Enable

| | Details | Address | Login Name | First Name | Last Name | Location | IP Address | AST Device | Registered | | |
|--------------------------|---------|-----------------|-----------------|------------|-----------|-----------|------------------|--------------------------|--|--------------------------|--------------------------|
| | | | | | | | | | Prim | Sec | Surv |
| <input type="checkbox"/> | Show | 10067@sglab.com | 10067@sglab.com | One | IP652 | Location1 | 10.1.10.178:5060 | <input type="checkbox"/> | <input checked="" type="checkbox"/> (AC) | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | Show | 10068@sglab.com | 10068@sglab.com | Two | IP652 | Location1 | 10.1.10.161:5060 | <input type="checkbox"/> | <input checked="" type="checkbox"/> (AC) | <input type="checkbox"/> | <input type="checkbox"/> |

From the web interface of the IP652 phones, click **Status** → **Basic** from the top menu. Verify that the line status for **Line 1 Status** shows as **Online**.

VOIP-IP652 ... control panel

Firmware Version: 1.3.4 (Hy0811225428)
DSP Version: D2.63
Current Time: Oct 28 14:18:52 2011
Admin Mode [\[Logout\]](#)

Status SIP Account Network Phone Administration

Basic DHCP Syslog

Product Information

Product Information

| | |
|----------------------------|----------------------|
| Product Name: | IP652 |
| Internet(WAN) MAC Address: | 00:21:F2:04:2E:27 |
| PC(LAN) MAC Address: | 00:21:F2:04:2E:26 |
| Hardware Version: | 1.0.1 |
| Firmware Version: | 1.3.4 (Hy0811225428) |
| DSP Version: | D2.63 |

Line Status

Line Status

| | |
|----------------|---------|
| Line 1 Status: | Online |
| Line 2 Status: | Disable |
| Line 3 Status: | Disable |

Help

Product Information:
It shows the basic information of the product.

Line Status:
It shows the registration state of each line.

Network Status:
It shows the information of WAN port, VPN and LAN port.

System Status:
It shows the current time and the running time of the product.

9. Conclusion

These Application Notes describe the configuration steps required for Flying Voice Technology IP652 SIP Phones to successfully interoperate with Avaya Aura® Session Manager 6.1 and Avaya Aura® Communication Manager 6.0.1. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references documentation relevant to these Application Notes. Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.0, Doc ID 03-300509, June 2010.
- [2] *Administering Avaya Aura® Session Manager*, Release 6.1, Doc ID 03-603324, Issue 1, November 2010.

The following the App Notes will be shipped with the product by Flying Voice Technology.

- [3] *Flying Voice IP652 User Manual*, V1.9, May 2011,

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