Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager R6.2 and Avaya Aura® Communication Manager R6.2 to interoperate with Zenitel Stentofon - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Zenitel Stentofon to interoperate with Avaya Aura® Session Manager R6.2 and Avaya Aura® Communication Manager R6.2. The Zenitel Stentofon is a door IP Intercom that supports voice transmission using the Session Initiation Protocol (SIP).

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 configuration steps required for Zenitel Stentofon IP Intercom Substation to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The Zenitel Stentofon IP Intercom Substations is a door communicator that supports voice transmission using the Session Initiation Protocol (SIP), in addition to being a door entry device. In the compliance testing, the Zenitel Stentofon IP Intercom Substation was set up as a SIP user on Avaya Aura® Session Manager and underwent testing of various call scenarios with other Avaya telephones and Zenitel Stentofon IP Intercom Substations.

2. General Test Approach and Test Results

The general test approach was to place calls to and from Stentonfon and exercise basic telephone operations. For serviceability testing, failures such as cable pulls and hardware resets were performed.

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.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing was to verify that:

- Stentofon successfully registers with Session Manager using IP address and FQDN.
- Stentofon successfully establishes audio calls with Avaya H.323, SIP and digital endpoints registered to Session Manager and Communication Manager.
- Stentofon successfully establishes audio calls with PSTN.
- Stentofon IP successfully negotiates the appropriate audio codec.
- DTMF tones could be passed successfully to energize relay on Stentofon unit and switch audio direction.
- Stentofon successfully calls multiple destinations using a cover answer group.
- Stentofon successfully calls a variety of endpoints in its call list.
- Correct handling of forwarded calls, cover paths and cover answer groups.

The serviceability testing focused on verifying the ability of Stentofon to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the devices and denying service on Session Manager.

2.2. Test Results

All test cases passed successfully with the following observation:

- When an Avaya 9640 SIP endpoint is placed in the calling list and a call is placed from Stentofon to the calling list, the invite contains the parameter “answer-mode=manual”. The Avaya SIP endpoint automatically answers the call. This is a bug in the Avaya SIP firmware.
- Stentofon was unable to register with Session Manager using FQDN.

2.3. Support

Technical support on Zenitel Stentofon can be obtained through the following:

- **Phone:** +47 4000 2700
- **Web:** http://www.zenitel.com/en/Stentofon/Service/

3. Reference Configuration

Figure 1 illustrates a test configuration that was used to compliance test the interoperability of Stentofon with Session Manager and Communication Manager. The configuration consists of Communication Manager, System Manager and Session Manager.Communication Manager has connections to a 9630 IP (H323) deskphone and 2420 Digital Telephone. Session Manager has SIP registrations with, Stentofon and 9640 IP (SIP) deskphone. An ISDN-PRI trunk connects Communication Manager to the PSTN.

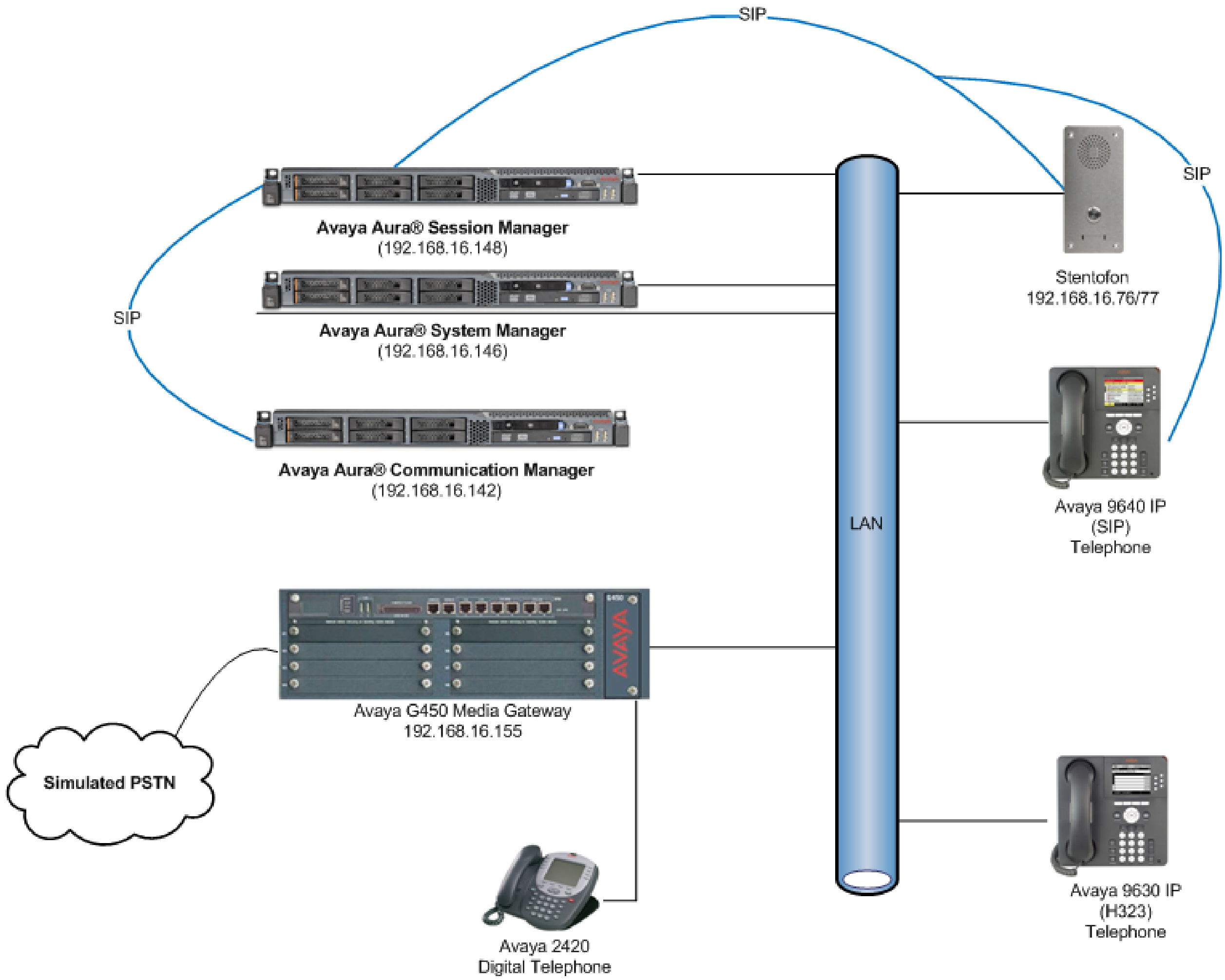


Figure 1: Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Zenitel Stentofon Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.2 SP1
Avaya Aura® System Manager running on Avaya S8800 Server	R6.2 SP1
Avaya Aura® Communication Manager running on Avaya S8800 Server	R6.2 SP1
Avaya 9630 IP Telephone	3.1 SP4 (H.323) 2.6.7 (SIP
Avaya 2420 Digital Telephone	N.A.
Zenitel Stentofon	Software version: 02.03.3.2

5. Configure Avaya Aura® Communication Manager

The configuration changes in this section for Communication Manager are performed through the Site Administration tool and via the System Manager web interface. Except where stated, the parameters in all steps are the default settings and are supplied for reference. For all other provisioning information such as provisioning of the trunks, call coverage, extensions, and voicemail, please refer to the Avaya product documentation in **Section 9.**
The procedures fall into the following areas:

- Configure IP Codec Set
- Configure SIP User
- Configure Endpoints for IP Video

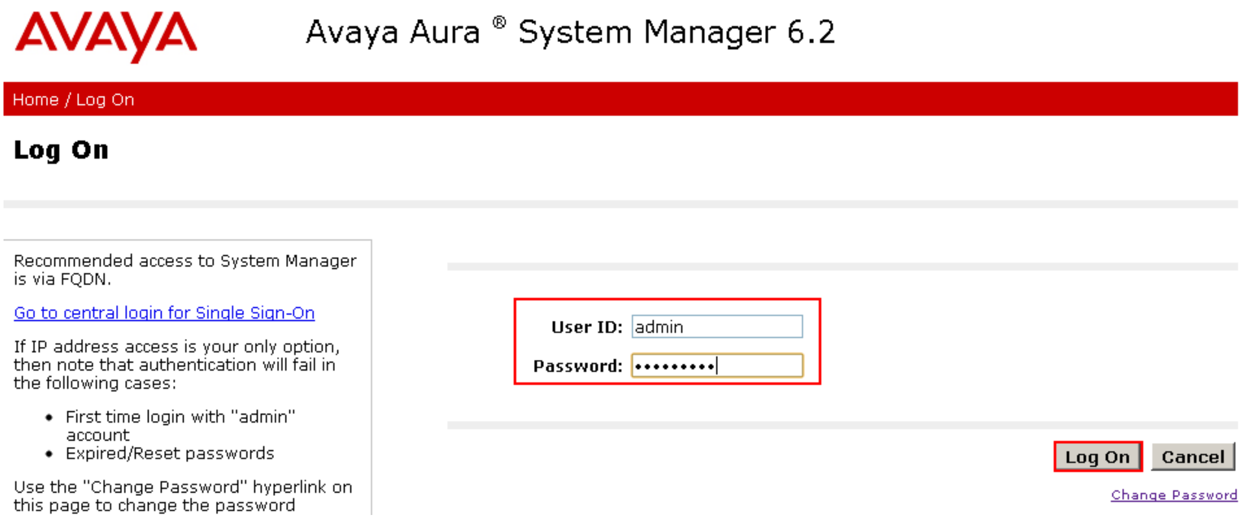
5.1. Configure IP Codec Set

The IP Codec set must be configured with the codecs for use by IP endpoints. Enter the command **change ip-codec-set x** where **x** is the relevant codec set and set the **Audio Codec** to be used on **Page 1**. In the example below, codecs **G.722-64K** and **G.711A** are configured.

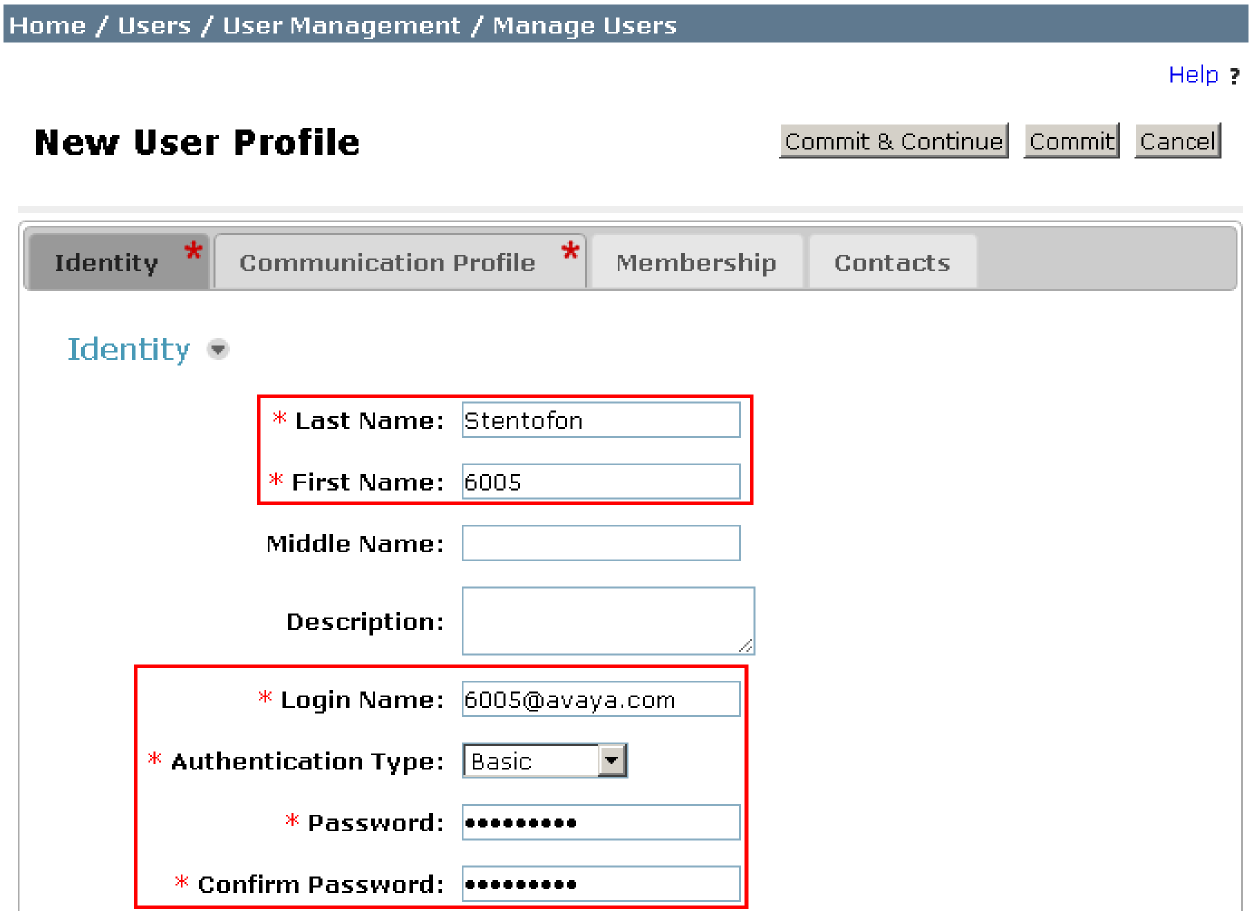
```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.722-64K                    2        20
 2: G.711A             n         2        20
 3:
 4:
 5:
 6:
 7:
```

5.2. Configure SIP User

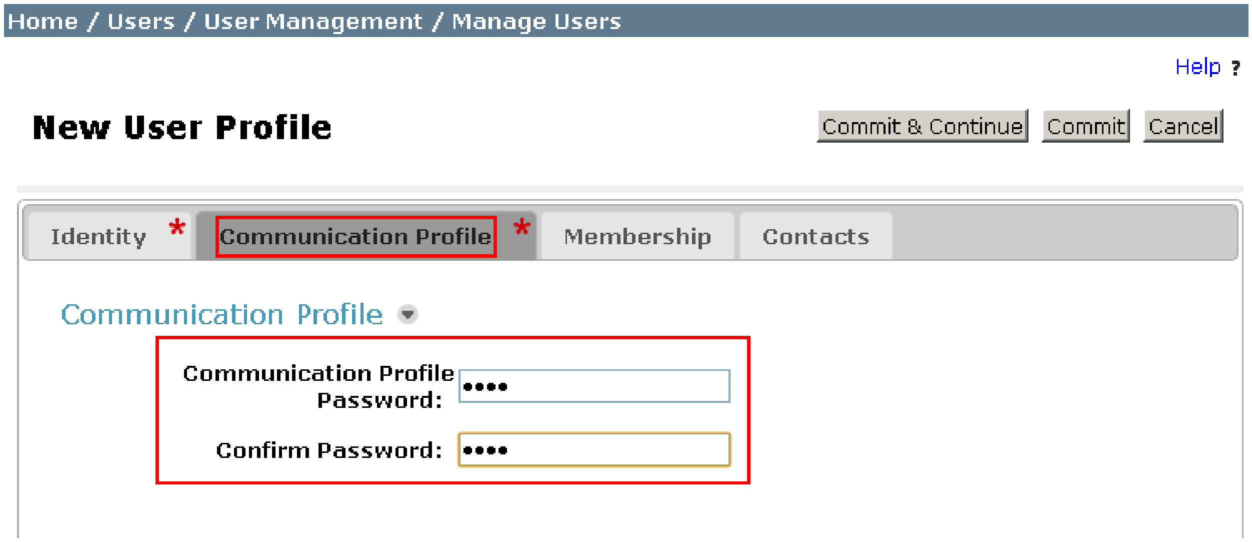
A SIP user must be added for each Stentofon endpoint required. Navigate to the System Manager web interface, in this case **https://192.168.16.146/SMGR** and login with the relvant credentials.



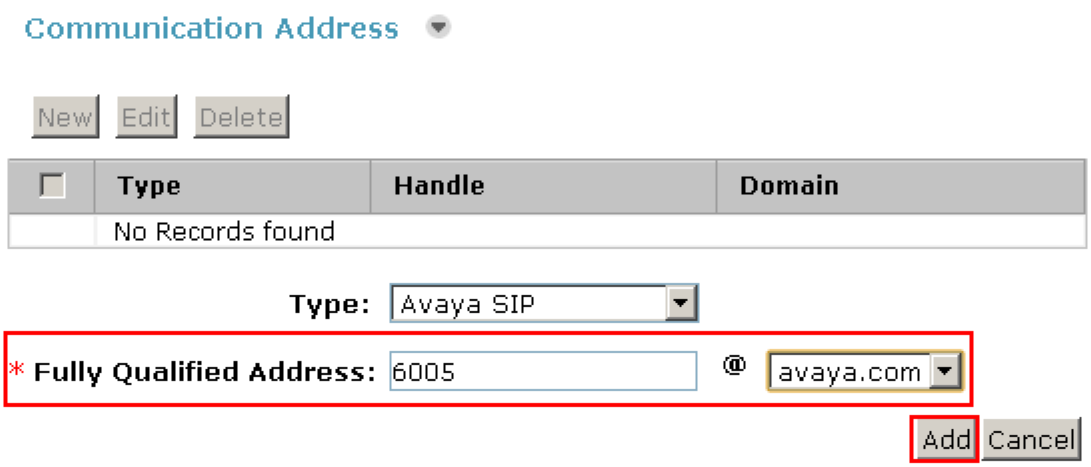
Click **User Management → Manage Users → New** enter an identifying **Last Name** and **First Name**, enter an appropriate **Login Name**, set **Authentication Type** to **Basic** and administer a password in the **Password** and **Confirm Password** fields.



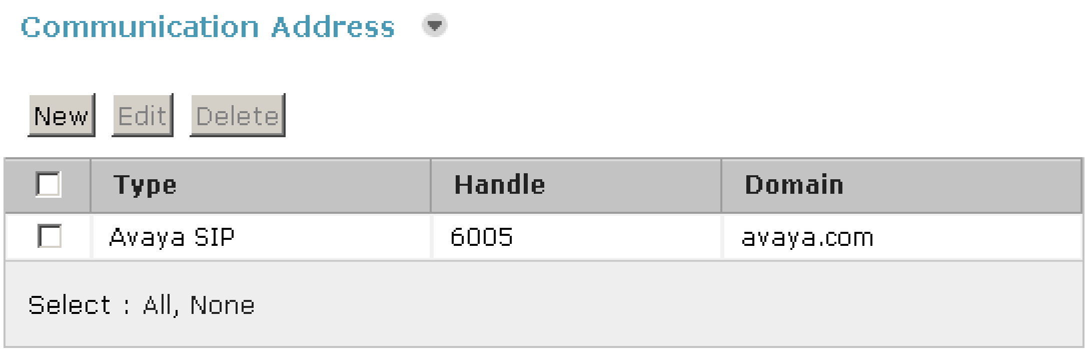
Click on the **Communication Profile** tab and enter and confirm a **Communication Profile Password**, this is used when logging in the SIP endpoint.



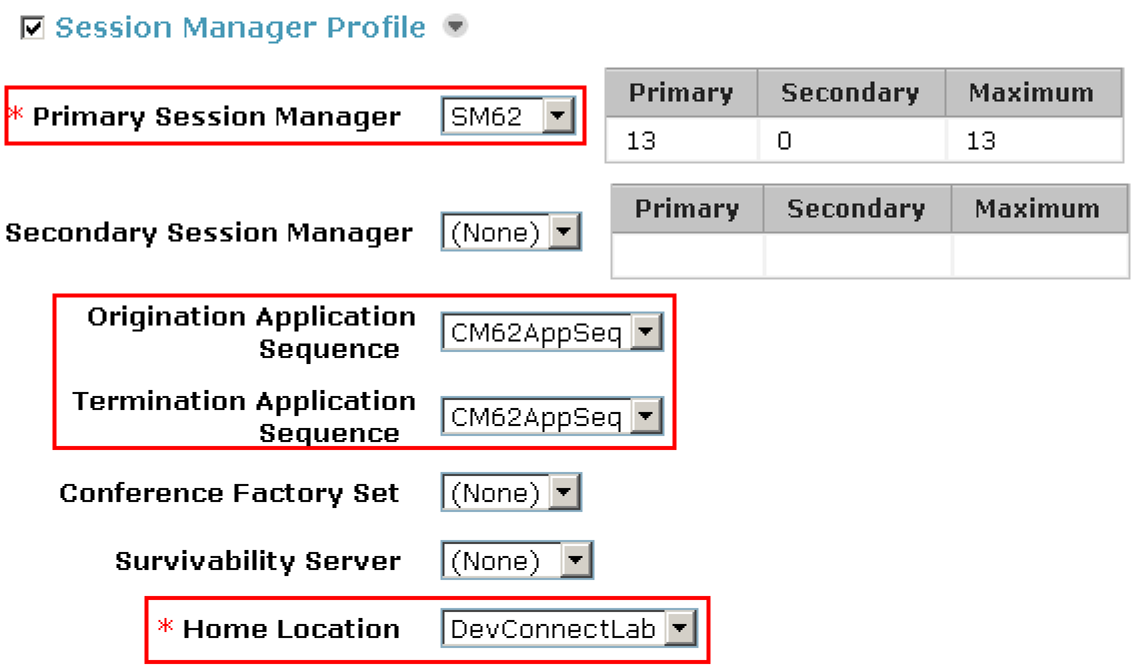
On the same page, scroll down and under **Communication Address** click **New**, select **Avaya SIP** from the **Type** drop down box and enter the **Fully Qualified Address** of the new SIP user. Click **Add** when done.



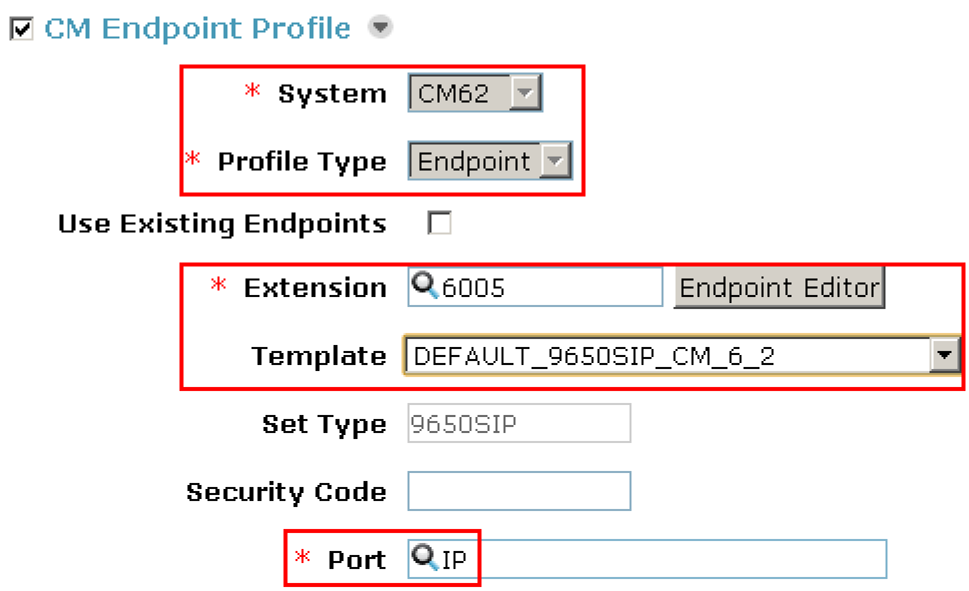
The Communication Address will now appear added.



Continue to scroll down on the same page, enter the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location** relevant to the implementation.



Scroll down to the page to the **CM Endpoint Profile** section. Select the Communication Manager system from the **System** drop down box, select **Endpoint** as the **Profile Type**, enter the **Extension** number you wish to use, select **DEFAULT_9650SIP_CM_6_2** as the **Template** and ensure **IP** is configured as the **Port**, click Commit (not shown) when done. Repeat this for every SIP extension required.



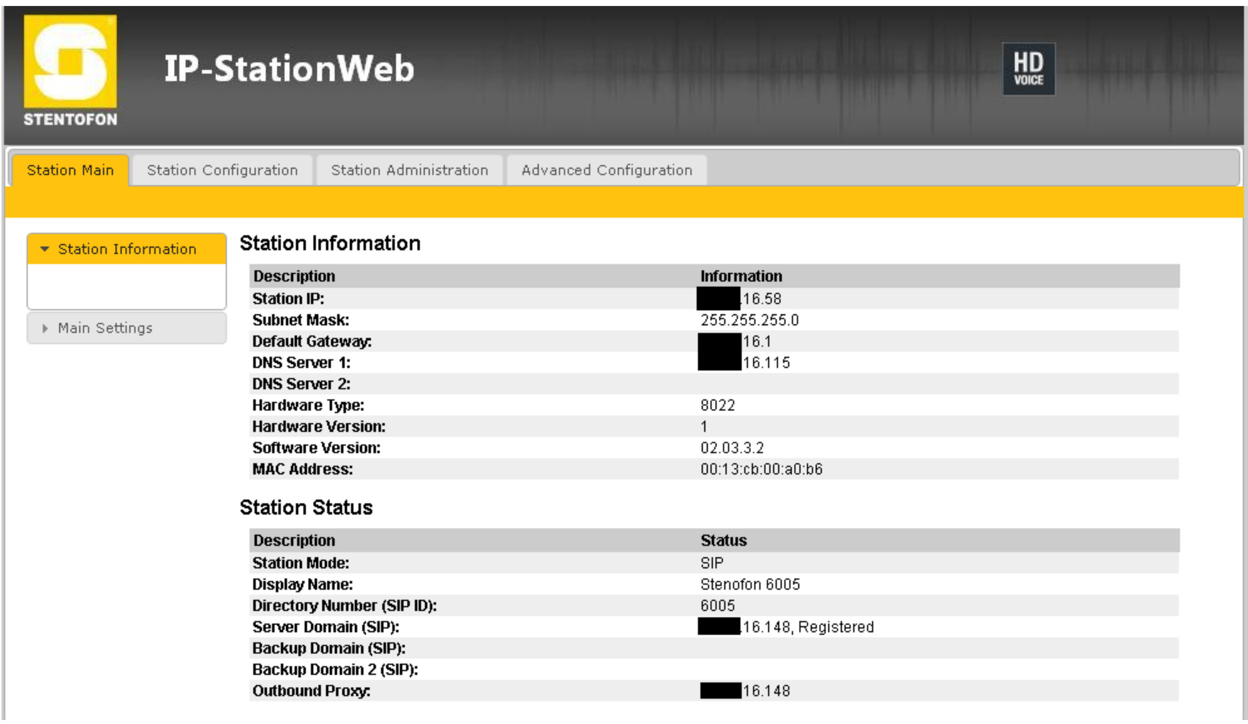
6. Configure Zenitel Stentofon

The following steps detail the configuration for Stentofon using the Web Interface. The steps include the following areas:

- Launch Web Interface
- Administer SIP Settings
- Configure Direct Access Key

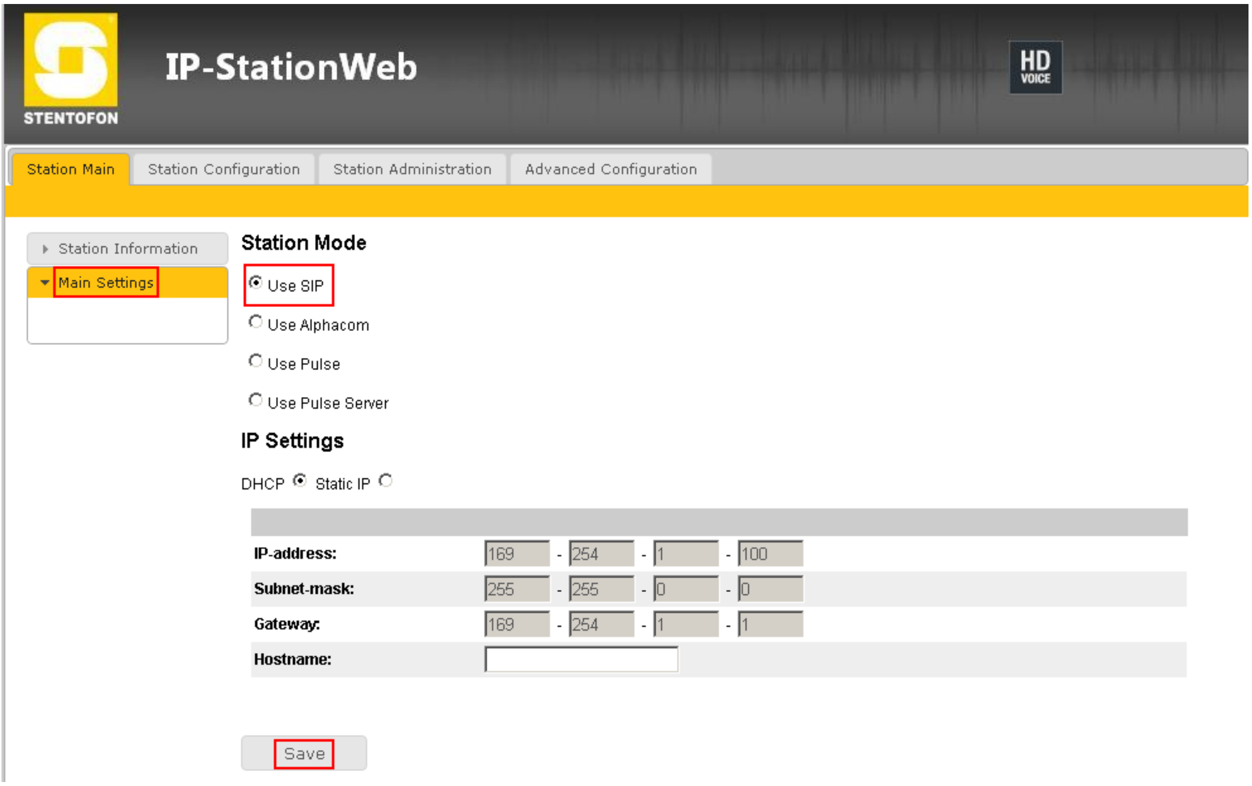
6.1. Launch Web Interface

Access the Stentofon web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of Stentofon. Log in with the appropriate credentials. The **IP-StationWeb** screen is shown.



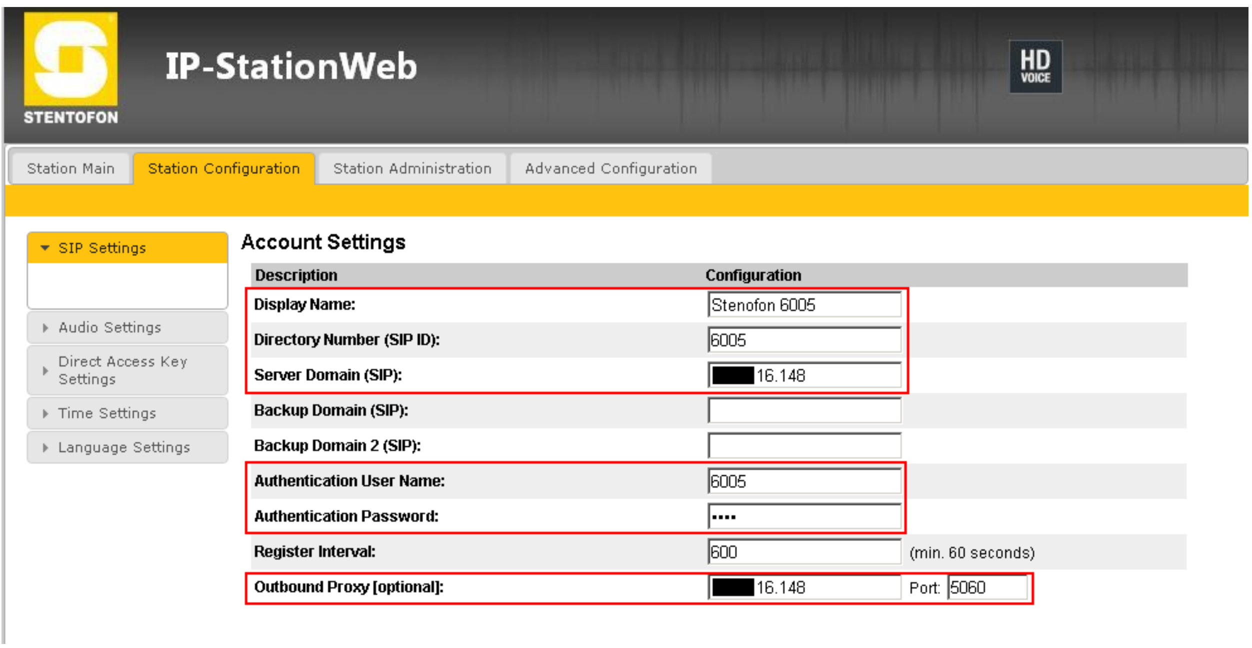
6.2. Administer SIP Settings

Select **Main Settings** from the left menu and select **Use SIP**, click **Save** when done. A screen will appear (not shown) to confirm the setting, click Apply and Stentofon will reboot.

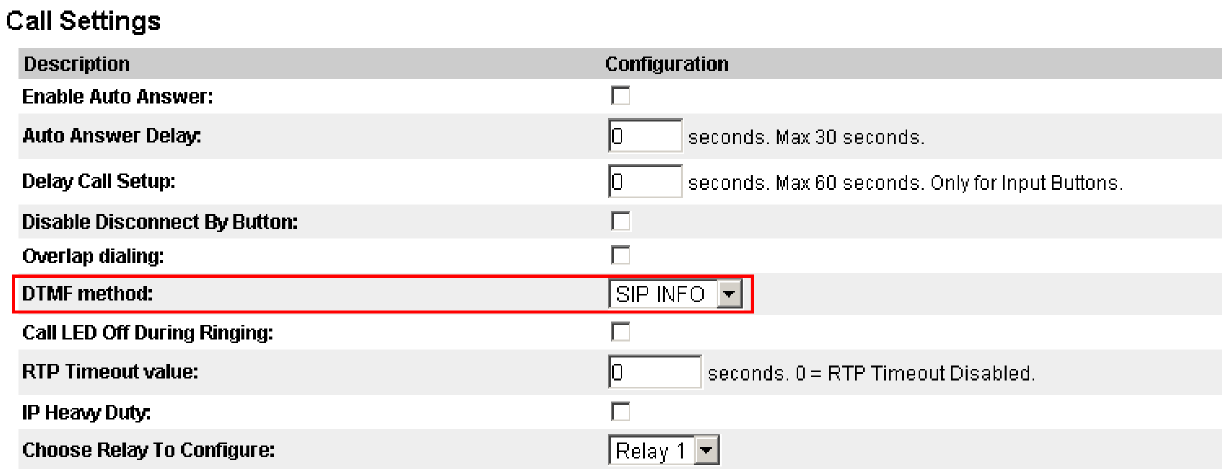


Click on **Station Configuration → SIP Settings** and configure the following in the **Account Settings** section:

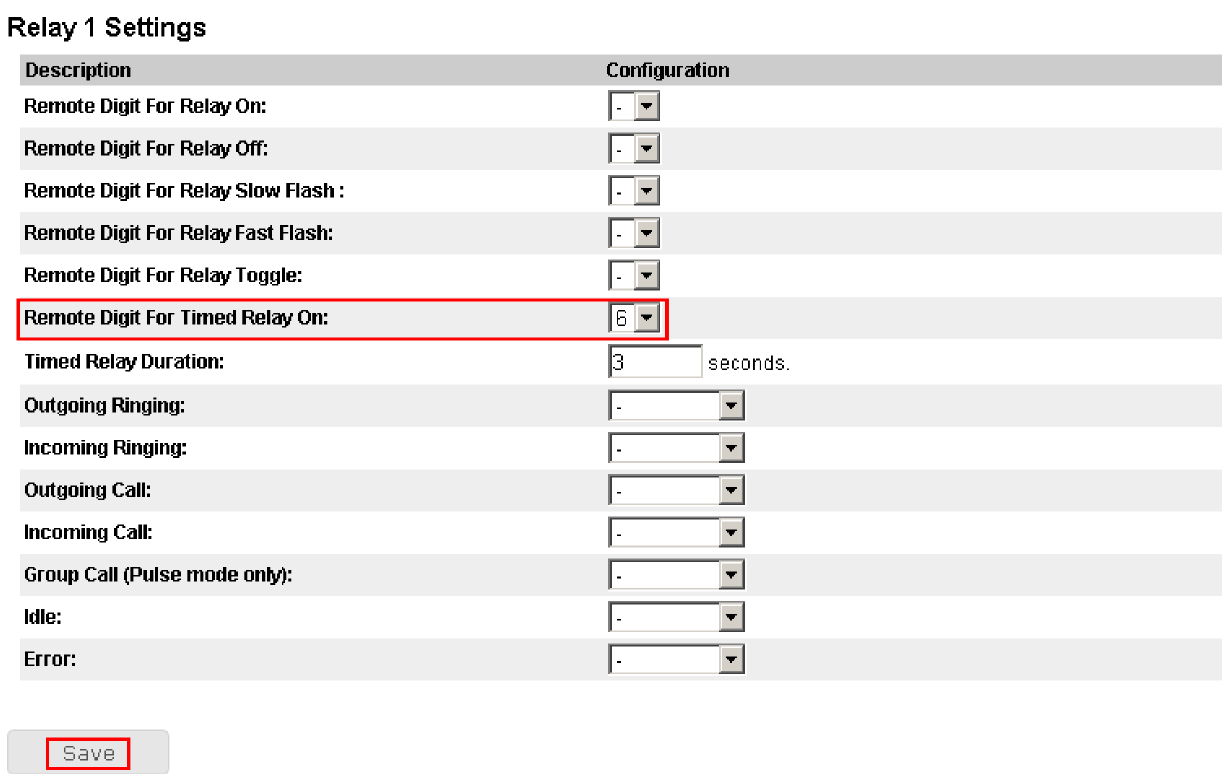
- **Display name**: Enter the desired name.
- **Directory Number (SIP ID**: Enter a user extension administered from **Section 5.2**.
- **Server Domain (SIP)**: Enter the IP address of Session Manager.
- **Authentication User Name**: Enter a user extension administered from **Section 5.2**.
- **Authentication Password**: Enter the **Communication Profile Password** from **Section 5.2**.
- **Outbound Proxy (optional)**: Enter the IP address of Session Manager and **5060** as the **Port.**



In the **Call Settings** section, configure as required the **DTMF Method** as **SIP INFO** or RFC 2833 (not shown), this allows DTMF tones to be either sent in-band or using SIP INFO messaging. Configure other options as required.

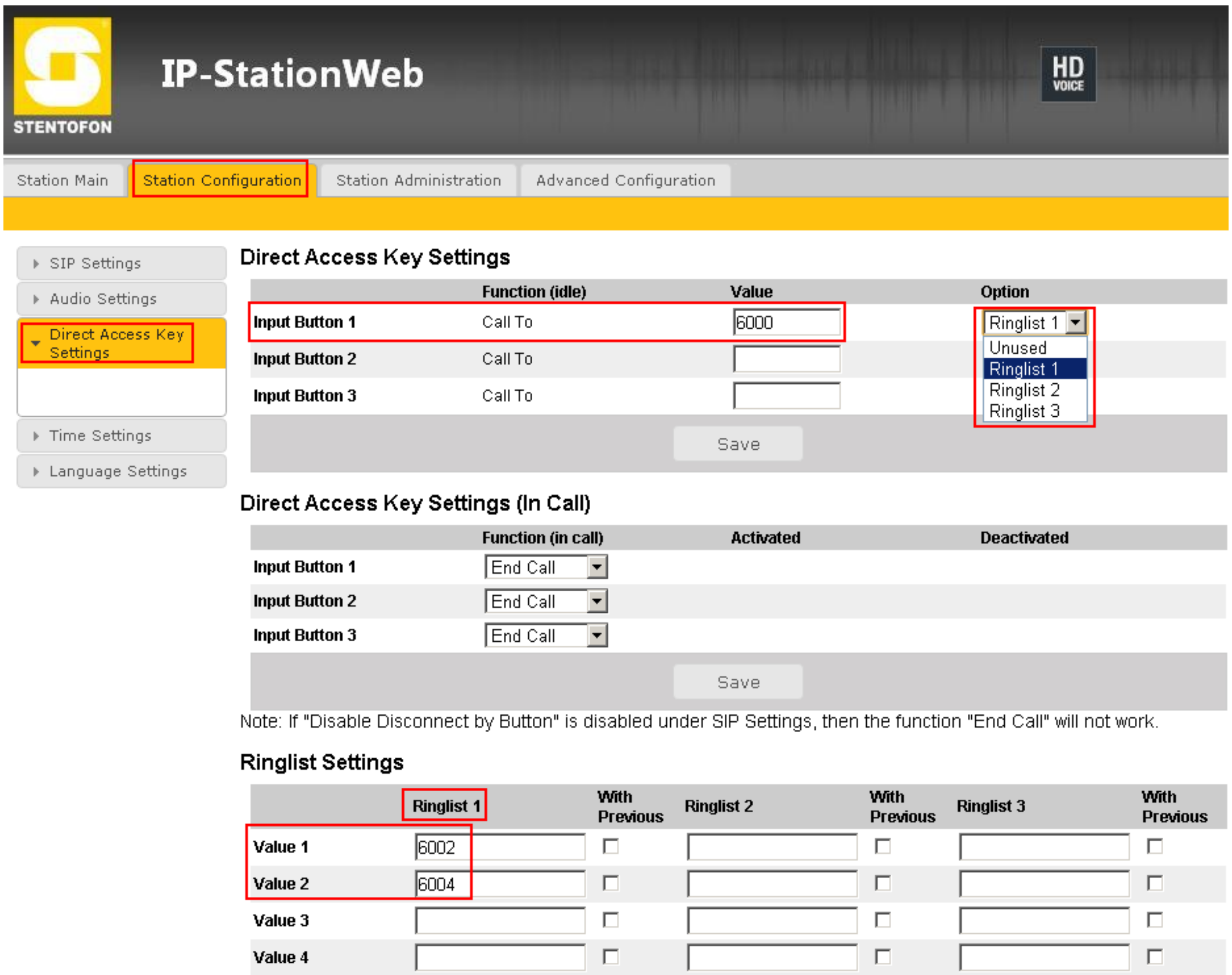


In the **Relay 1 Settings** section enter select a digit from the drop down box for **Remote Digit for Timed Relay On**. When this digit is pushed by a called party, the relay in the Stentofon will be energized. Retain the default values for the remaining fields. Click **Save** when done. A screen will appear (not shown) to confirm the setting, click Reboot and Stentofon will reboot



6.3. Configure Direct Access Key

Select **Station Configuration → Direct Access Key Settings** from the left menu and select one of the buttons (e.g. 1 to 3 as shown below) to configure it. In the case of the compliance test, the unit tested had only Direct Access Key. In the **Value** field enter the extension to be called when the Direct Access Key is pushed, choose **Unused** from the **Option** drop down box. If **Ringlist 1** is chosen, configure the **Ringlist Settings** and enter an extension number to be dialed in the **Value 1, Value 2** etc fields. In this example **Ringlist 1** is configured to call **6002** and then **6004**, select the **With Previous** tick box in order for the first extension to continue ringing when the second extension starts to ring. Click Save (not shown) when done.

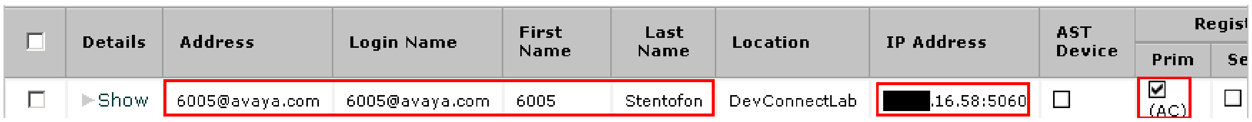


7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Session Manager and Stentofon.

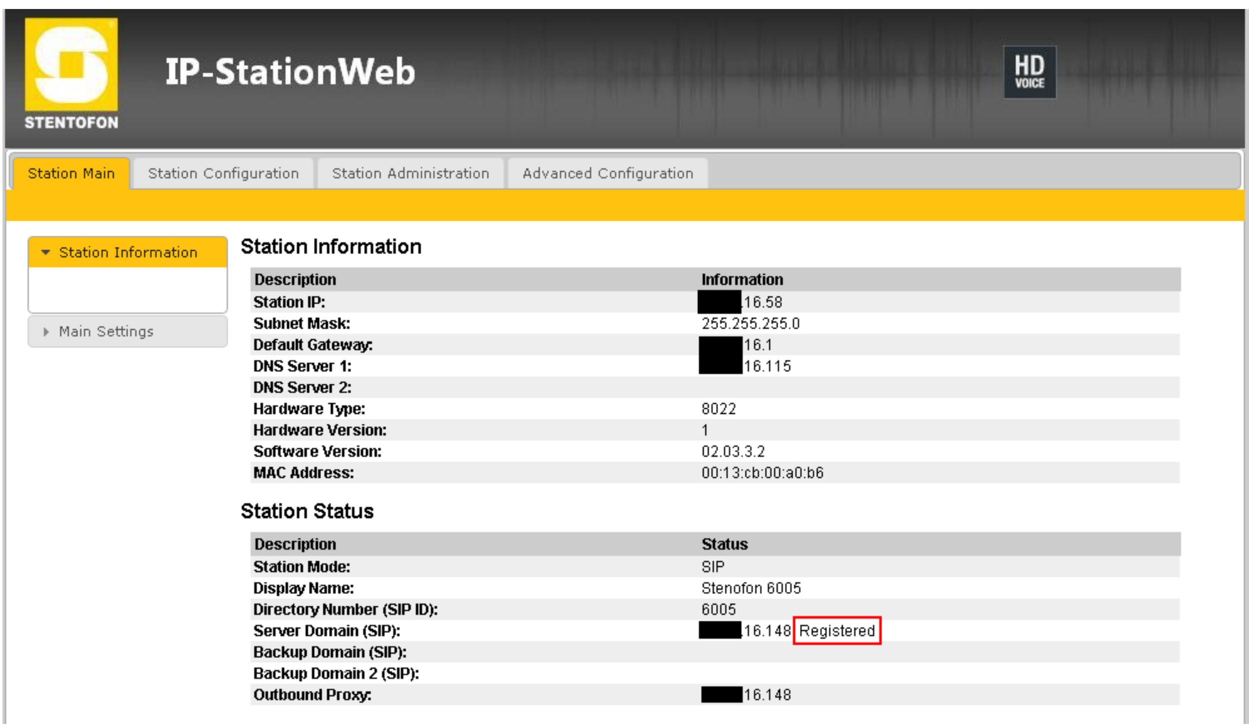
7.1. Verify Avaya Aura® Session SIP Endpoint Registration

From the System Manager web interface click **Session Manager → System Status → User Registrations.** Verify that Stentofon endpoints are successfully registered as shown below.



7.2. Verify Stentofon SIP Registration

From the Stentonfon web interface, select **Information** from the left menu. Verify that the **Registration state** shows **Registered**. Place a call to another endpoint to verify basic call operation.



7.3. Verify Succesful Calls

Place a call to and from the Stentofon endpont. Verify 2-way audio is heard and validate call terminates successfully.

8. Conclusion

These Application Notes describe the configuration steps required for configuring Zenitel Stentofon to interoperate with Avaya Aura® Session Manager. All feature and serviceability tests were completed successfully with observations made in **Section 2.2**.

9. Additional References

This section references the Avaya and Zenitel product documentation that are relevant to these Application Notes.

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

[1] *Administering Avaya Aura® Session Manager Release 6.2, July 2012- Document Number 03-603324*.

The Zenitel Stentofon documentation can be found at http://www.zenitel.com.

[2] A100K11013-Pulse-Getting-Started.pdf.

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.