Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Server 1000E R7.6 to interoperate with Zenitel Stentofon Turbine Intercom using SIP Line Gateway- Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Zenitel Stentofon Turbine Intercom to interoperate with Avaya Communication Server 1000E R7.6 using SIP Line Gateway. The Zenitel Stentofon Turbine Intercom 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 Turbine Intercom to interoperate with Avaya Communication Server 1000E R7.6 using SIP Line Gateway (SLG). The Zenitel Stentofon Turbine Intercom 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 Turbine Intercom was set up as a 3rd Party SIP Extension on Avaya Communication Server 1000E and underwent testing of various call scenarios with other Avaya telephones and Zenitel Stentofon Turbine Intercom units.

2. General Test Approach and Test Results

The general test approach was to place calls to and from the Stentonfon endpoint 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 carried out to verify that:

- Stentofon successfully registers with SLG using IP address and FQDN.
- Stentofon successfully establishes audio calls with unistim and SIP and SIP connected to CS1000.
- Stentofon successfully establishes audio calls with PSTN.
- Stentofon successfully negotiates the appropriate audio codec.
- DTMF tones could be passed successfully to energize the relay on the Stentofon unit and switch audio direction.
- Stentofon successfully calls multiple destinations.
- Stentofon successfully calls a variety of endpoints in its Address Book.
- Correct handling of forwarded calls and call pickup.

The serviceability testing focused on verifying the ability of Stentofon to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the unit.

2.2. Test Results

All test cases passed successfully.

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 CS 1000E. The CS1000E R7.6 runs on the Common Processor Pentium Mobile (CPPM) server as a co-resident configuration. The SLG application on the signaling server co-resides on the CPPM. Element Manager is used to access the SLG which resides on the Unified Communication Management Server which is accessed through the System Manager. SIP and Unistim Avaya 1140 IP Deskphones were configured. Stentofon Turbine is registered to the SLG as a Third Party SIP Client (SIP3). An ISDN-PRI trunk connects the CS1000E to the PSTN.

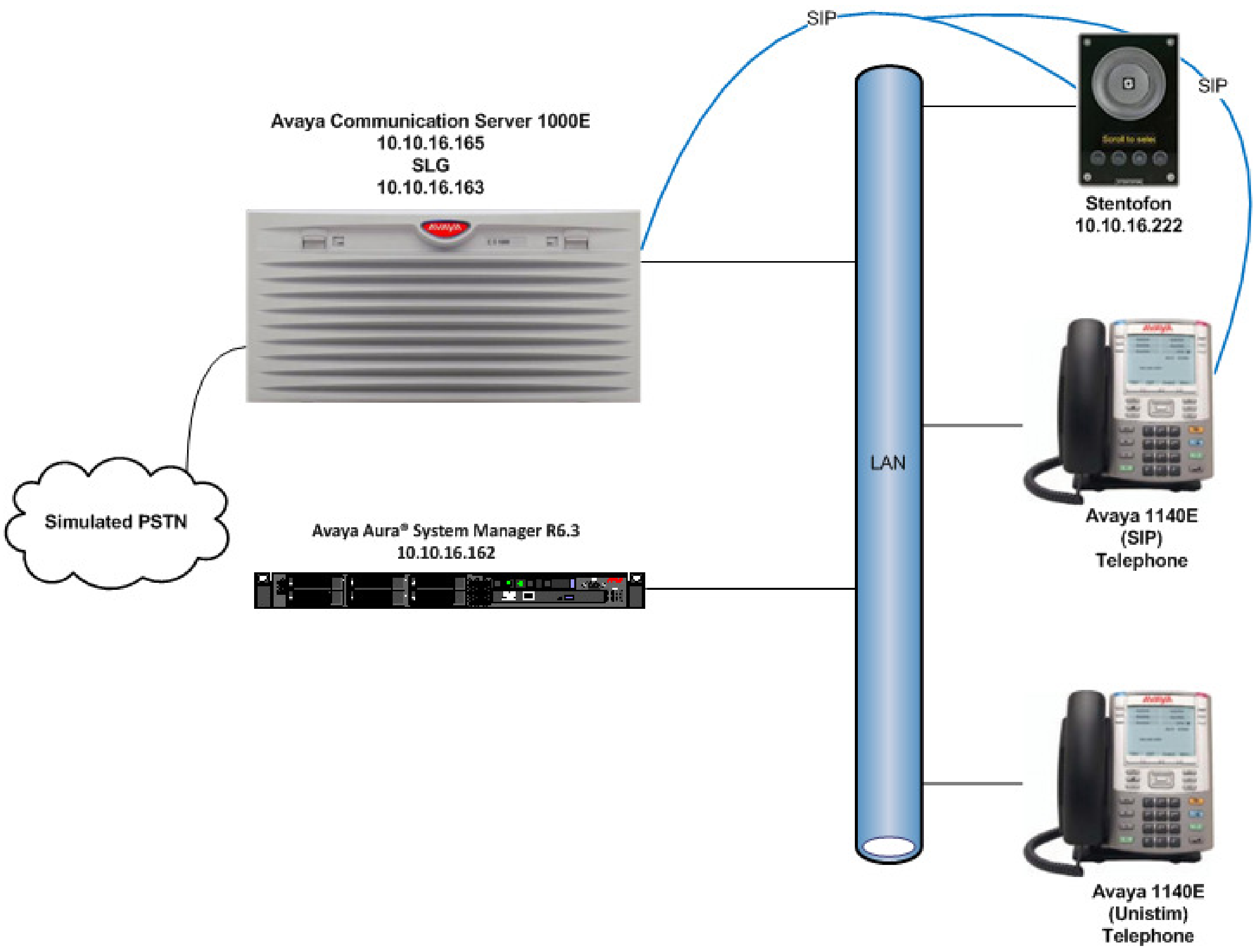


Figure 1: Avaya Communication Server 1000E and Zenitel Stentofon Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000E	7.6
Avaya 1140E IP Telephone	SIP:4.3 SP1 Unistim: 0625C8Q
Zenitel Stentofon Turbine Intercom	Software version: 02.03.3.2

5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the CS1000E over an SSH session. It is implied a working system is already in place and the SIP Line Gateway (SLG) is fully configured. Information regarding pre-configured components is obtained through the use of the Element Manager web interface. For all other provisioning information such as installation and configuration, please refer to the product documentation in **Section 9**.

5.1. Configuring Data block: SLS (SIP Line Services)

Create an **SLS Data** block using the **change** command in Overlay 15**.** Type **LD 15** to enter the overlay. The User Agent Prefix (**UAPR**) is required when configuring the **UEXT** for each Stentofon Turbine.

LD 15

Prompt	Response	Description
>**LD 15**		Enter Overlay 15
REQ	**CHG**	Change
TYPE	**SLS_DATA**	SIP Line Services Data block
CUST	**0**	Customer Number
SIPL_ON	YES	SIP Line on
UAPR	27	Prefix used to auto-generate the User Agent
NMME	NO	Multimedia Service

Note: This can also be accessed through Element Manager by clicking **Element Manager → Customer → Customer Details →SIP Line Service.**

5.2. Configuring Universal Extension (UEXT)

Configure the **UEXT** on the CS1000E using the **NEW** command in overlay 11. Type **LD 11** to enter the overlay. At the **Key 01** prompt use **UAPR** as configured in the **SLS_DATA Block** in **Section 5.1**. The SIP User (**SIPU**) and Station control Password (**SCPW**) are required when configuring each Stentofon Turbine.

LD 11

```
Prompt              Response            Description
>                   LD 11               Enter Overlay 11
REQ:                NEW                 Create New
TYPE:               UEXT                Universal Extension
TN                  100 0 02 00         Terminal Number
DES                 SIPL                Description
CUST                0                   Customer Number
UXTY                SIPL                Universal Extension type
MCCL                YES                 Maximum Client Count Limit
SIPN                0                   SIP Line for Nortel
SIP3                1                   SIP Line for third-party
FMCL                0                   Fixed Mobility Converged Line
SIPU                3020                Required for Stentofon Turbine USER ID
NDID                163                 Node ID
SUPR                NO                  Super User
ZONE                1                   Bandwidth Zone assigned for IP Sets
SCPW                1234                Required for Stentofon Turbine USER Password
KEY  00             SCR 3020 0          Key 0
CPND                New                 Calling Party Name Disply
Name                Stentofon2030       Name
Key  01             HOT U 273020        The HOT U number is derived from the UAPR as
configured in the SLS_DATA plus the Key 0 extension
Key  02
```

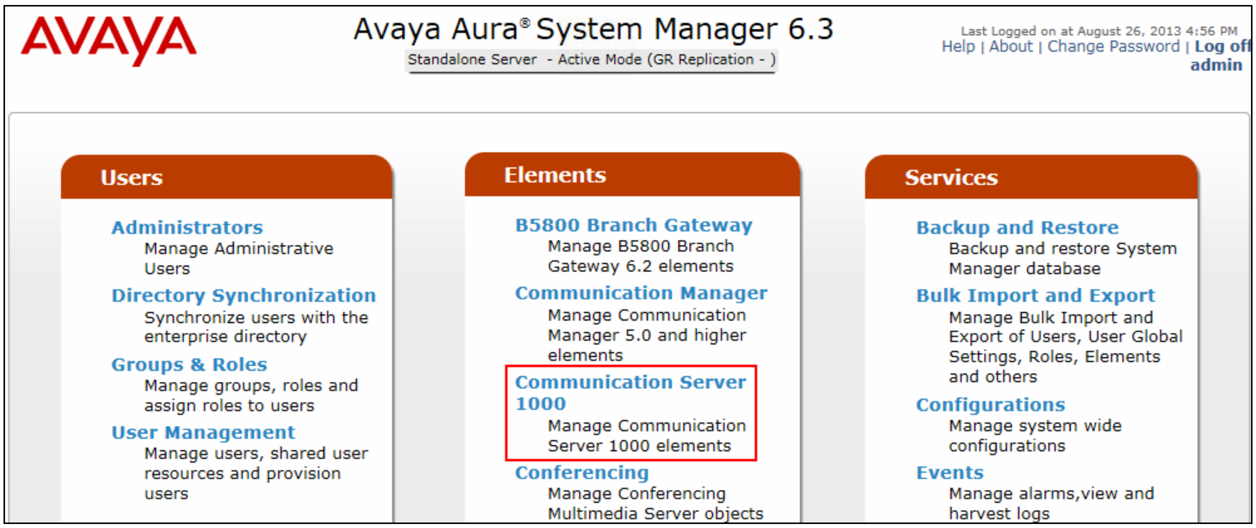
Note: This can also be configured through Element Manager click **Element Manager → Phones → Add**.

5.3. Obtain SIP Line information

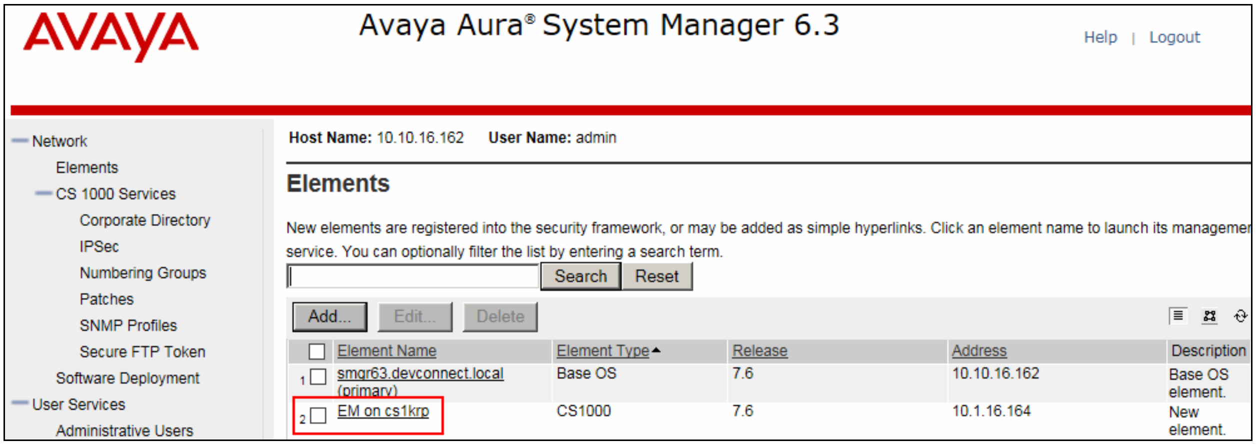
Certain information will be required in order to configure Stentofon Turbine. This information can be found using the Element Manager associated with the CS1000E. The following information can found in this section:

- **Telephony LAN (TLAN) Node IP Address**
- **SIP Domain Name**
- **SLG Local Sip Port**

Access the web GUI of the Unified Communication Management server, using the URL http://<fqdn>/SMGR, where "<fqdn> is the fully qualified domain name of System Manager. Log in with the appropriate credentials. Once logged in click **Communication Server 1000**.

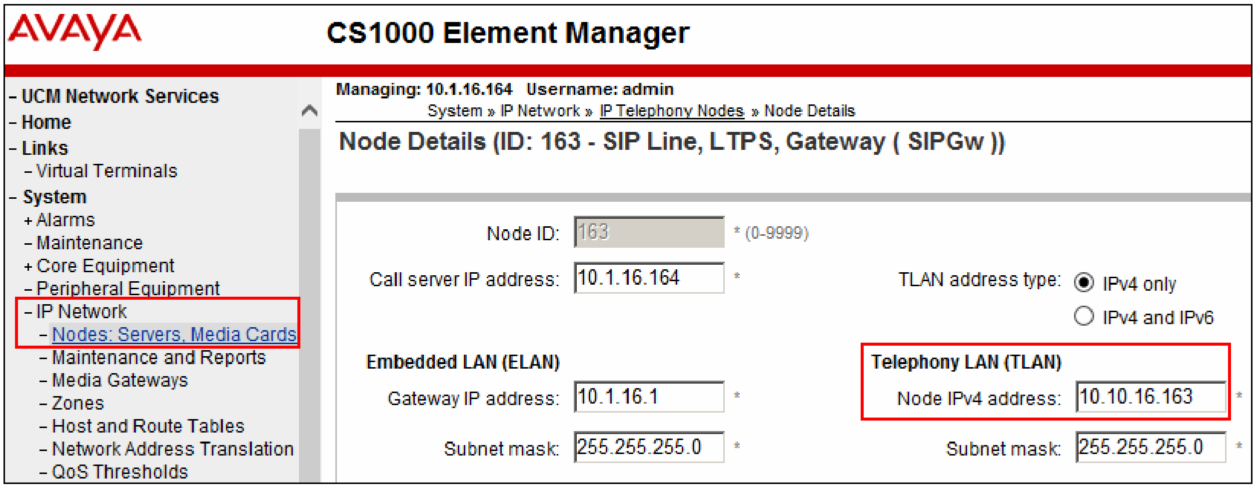


On the **Elements** page of UCM Services, select **Element Name** associated with the CS1000E. In this example it is **EM on cs1krp**.



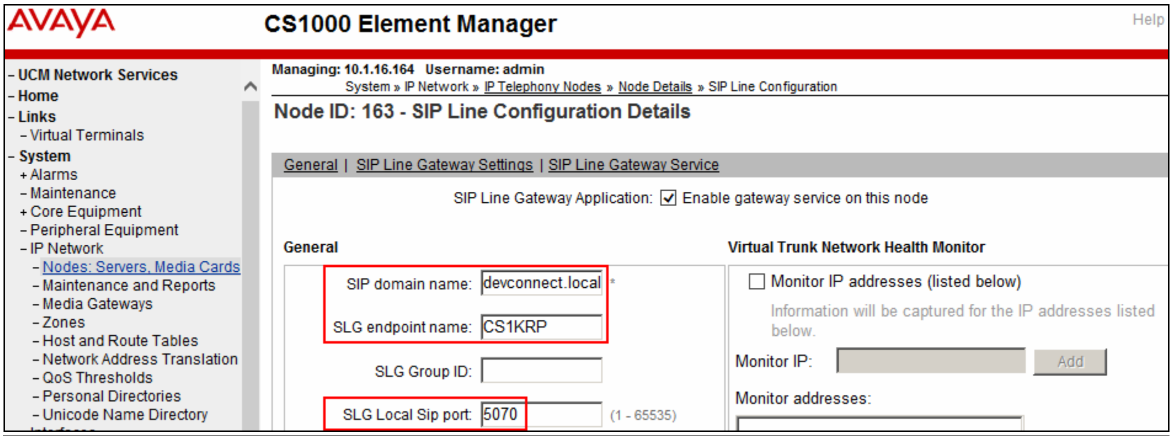
5.3.1. Obtain the Telephony LAN (TLAN) Node IP Address

Once the CS1000 Element Manager page opens, double click on the required Node ID (not shown) select **IP Network →Nodes Servers Media Cards**. On this page, the **Telephony LAN (TLAN) Node IP Address** can be located for information needed when configuring the Stentofon Turbine.



5.3.2. Finding the SIP Domain Name and SLG Local SIP Port

Using the scroll bar on the right side of the page scroll down and select **SIP LINE** (not shown). On this page the **SIP Domain Name, SLG endpoint name** and **SLG Local SIP Port** can be located for information needed when configuring the Stentofon Turbine.



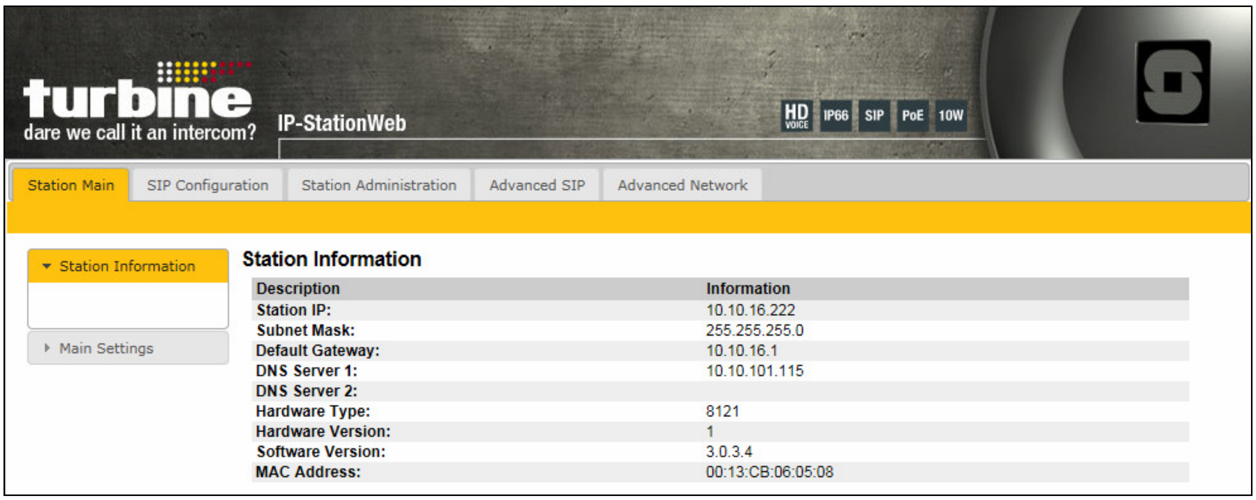
6. Configure Zenitel Stentofon Turbine Intercom

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

- Launch Web Interface
- Administer SIP Settings

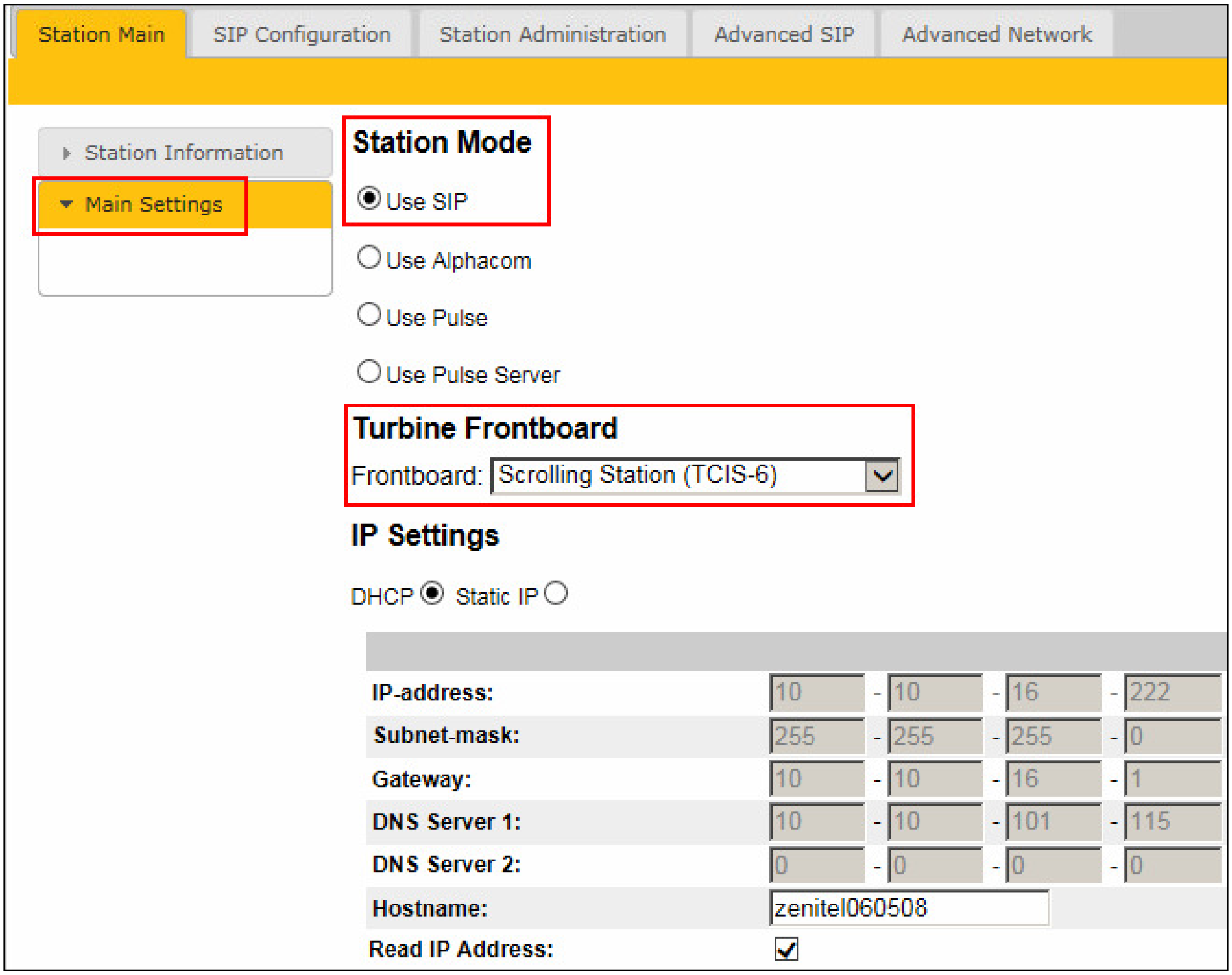
6.1. Launch Web Interface

Access the Stentofon web interface, enter **http://<ipaddress>** in an Internet browser window, where **<ipaddress>** is the IP address of the Stentofon Turbine unit. Log in with the appropriate credentials. The **IP-StationWeb** screen is shown. In this instance the unit has obtained its IP address via DHCP, for more information on configuring Stentofon Turbine, see **Section 9**



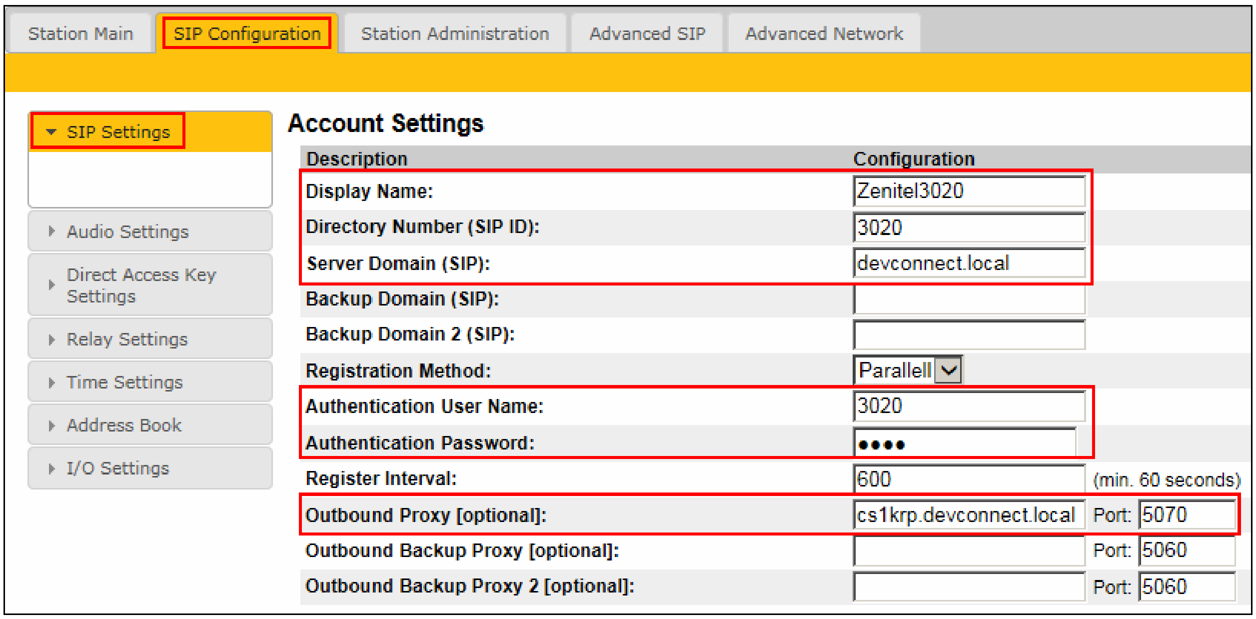
6.2. Administer SIP Settings

Select **Main Settings** from the left menu and select **Use SIP**, and select the appropriate device model from the **Turbine Frontboard** drop down list, in this case **Scrolling Station (TCIS-6)**. Click **Save** (not shown) when done. A screen will appear (not shown) to confirm the setting, click Apply and Stentofon will reboot.

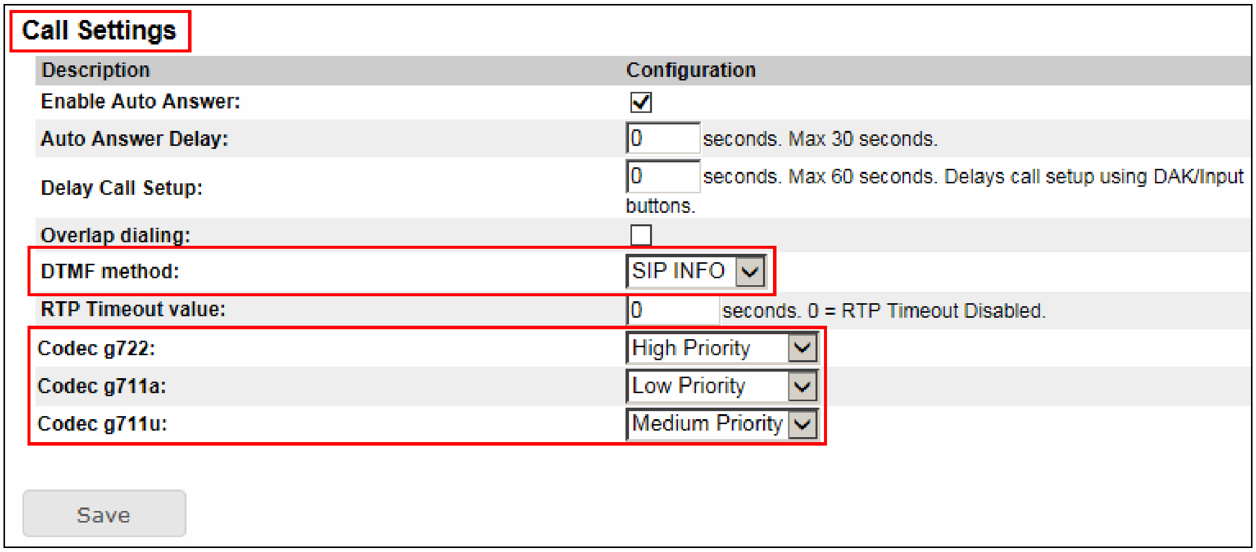


Click on **SIP Configuration → SIP Settings** and configure the following in the **Account Settings** section:

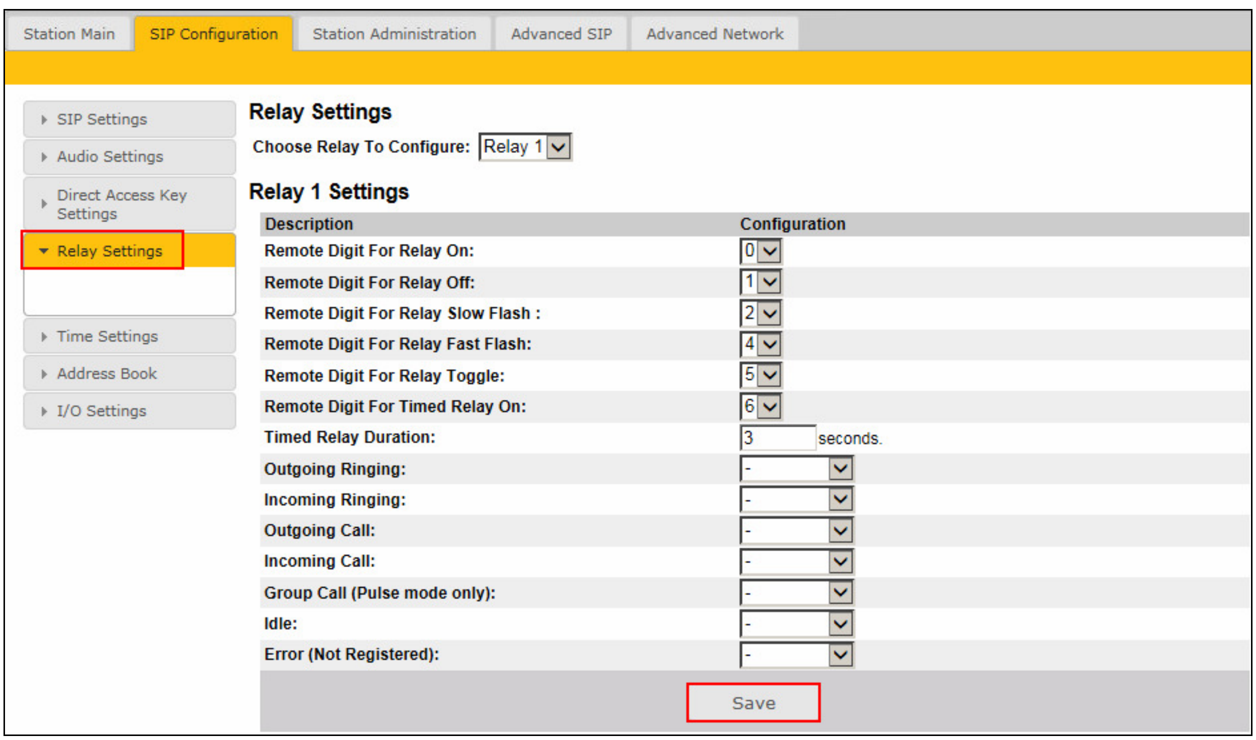
- **Display name**: Enter the desired name.
- **Directory Number (SIP ID)** Enter a user extension administered in **Section 5.2**.
- **Server Domain (SIP)**: Enter the domain name obtained in **Section 5.3.2**
- **Authentication User Name**: Enter a user extension administered from **Section 5.2**.(field **SIPU**)
- **Authentication Password**: Enter the **Communication Profile Password** from **Section 5.2**. (field **SCPW**)
- **Outbound Proxy (optional)**: Enter the FQDN of the SLG and **5070** as the **Port.**



In the **Call Settings** section, configure as required the **DTMF Method** as **SIP INFO** or RFC 2833 (not shown) and the coded priorities.



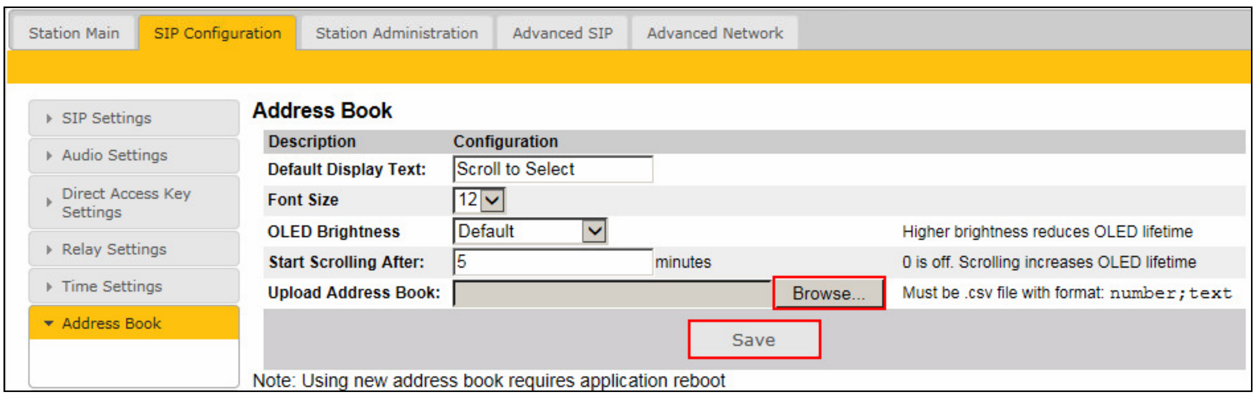
In the **Relay Settings** screen select a digit from the drop down box to activate the various Relay features. When this digit is pushed by a called party, the relay in the Stentofon will be energized/de-energized. Click **Save** when done. A screen will appear (not shown) to confirm the setting, click Reboot and Stentofon will reboot



In the **Address Book** screen click **Browse** and navigate to a pre-defined address book. This file must be in CSV format such as the following example.

```
3021;SIP Station 3021
3022;Other Stentofon 3022
3000;Unistim Station 3000
3001;Unistim Station 3001
3050;HuntGroup 3050
3023;SIP Station 3023
6000;PSTN6000
*88;Pickup *88
*13023;FWDto3023
*13001;FWDto3001
#1;FWDCancel
```

Click **Save** and reboot the application from the **Station Administration** tab when done.



7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of CS1000E and Stentofon.

7.1. Verify Avaya Communication Server 1000E SIP Station Registration

Check the status of the Stentofon Turbine SIP registration by opening an SSH session to the signaling server.

- Login with the appropriate credentials.
- At the prompt enter the following command **slgSetShowAll.**

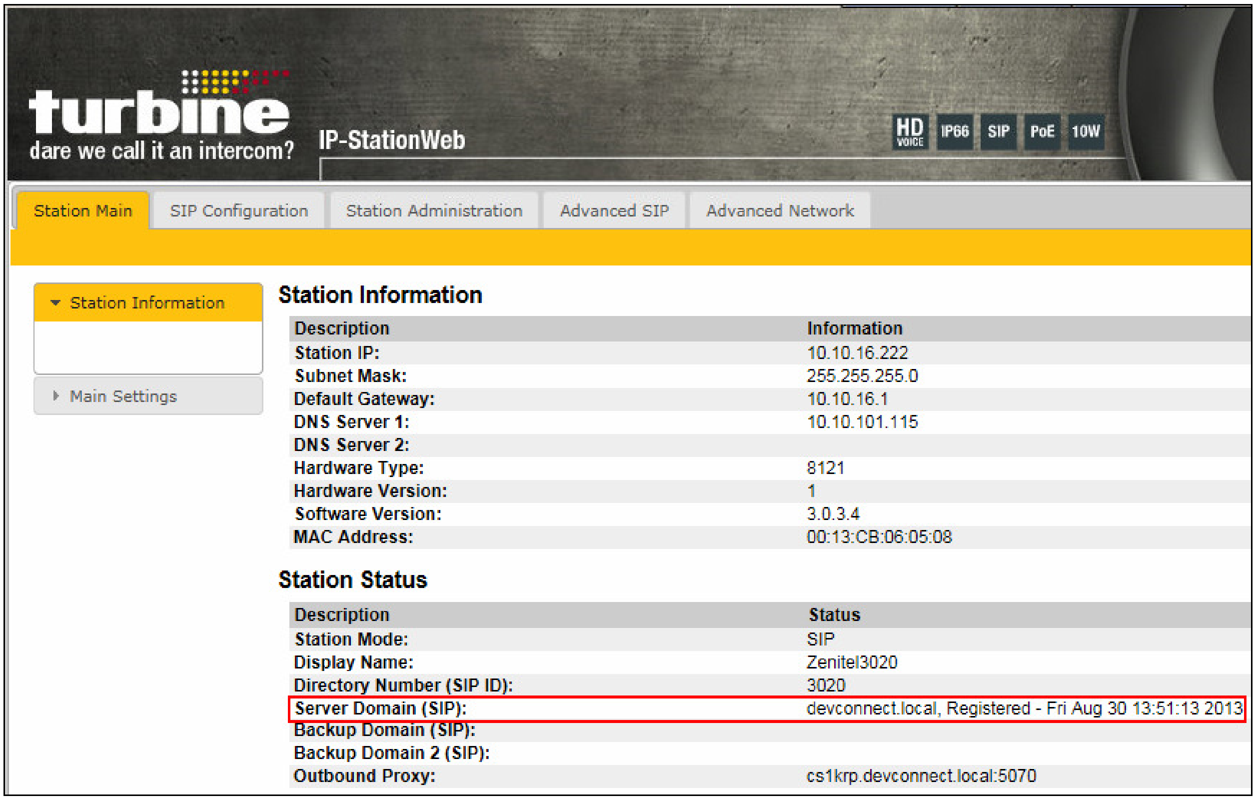
Example Below shows that the Stentofon Turbine 3020 is registered.

```
[admin@cs1krp ~]$ slgSetShowAll

=== VTRK ===
UserID           AuthId      TN           Clients  Calls  SetHandle  Pos ID    SIPL Type
---------------  ----------  -------------  -------  -----  ---------  -------   -----
---------------- IPV4 Endpoints   --------------------------
                3020        3020     100-00-02-00        1      0  0x9ea3f48   SIP Lines
Total User Registered = 1   V4 Registered = 1  V6 Registered = 0
```

7.2. Verify Stentofon SIP Registration

From the Stentonfon Turbine web interface, select **Station Information** from the left menu. Verify that the **Station Status Server Domain (SIP)** shows **Registered**. Place a call to another endpoint to verify basic call operation.



8. Conclusion

These Application Notes describe the configuration steps required for configuring Zenitel Stentofon Turbine to interoperate with Avaya Communication Server 1000E. All feature and serviceability tests were completed successfully with any 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 documentation that is relevant when administering the test configurations is outlined below. Product documentation for Avaya products is available at *http://support.avaya.com*.

[1] *Software Input Output Reference — Administration Avaya Communication Server 1000 Release 7.6 NN43001-611 Issue 06.02 April 2013*

[2] *SIP Line Fundamentals Avaya Communication Server 1000 Release 7.6 NN43001-508 Issue 04.01 March 2013*

[3] *System Management Reference Avaya Communication Server 1000 7.5 NN43001-600, 05.08 December 2011*

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

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.