



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 Stentofon 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.

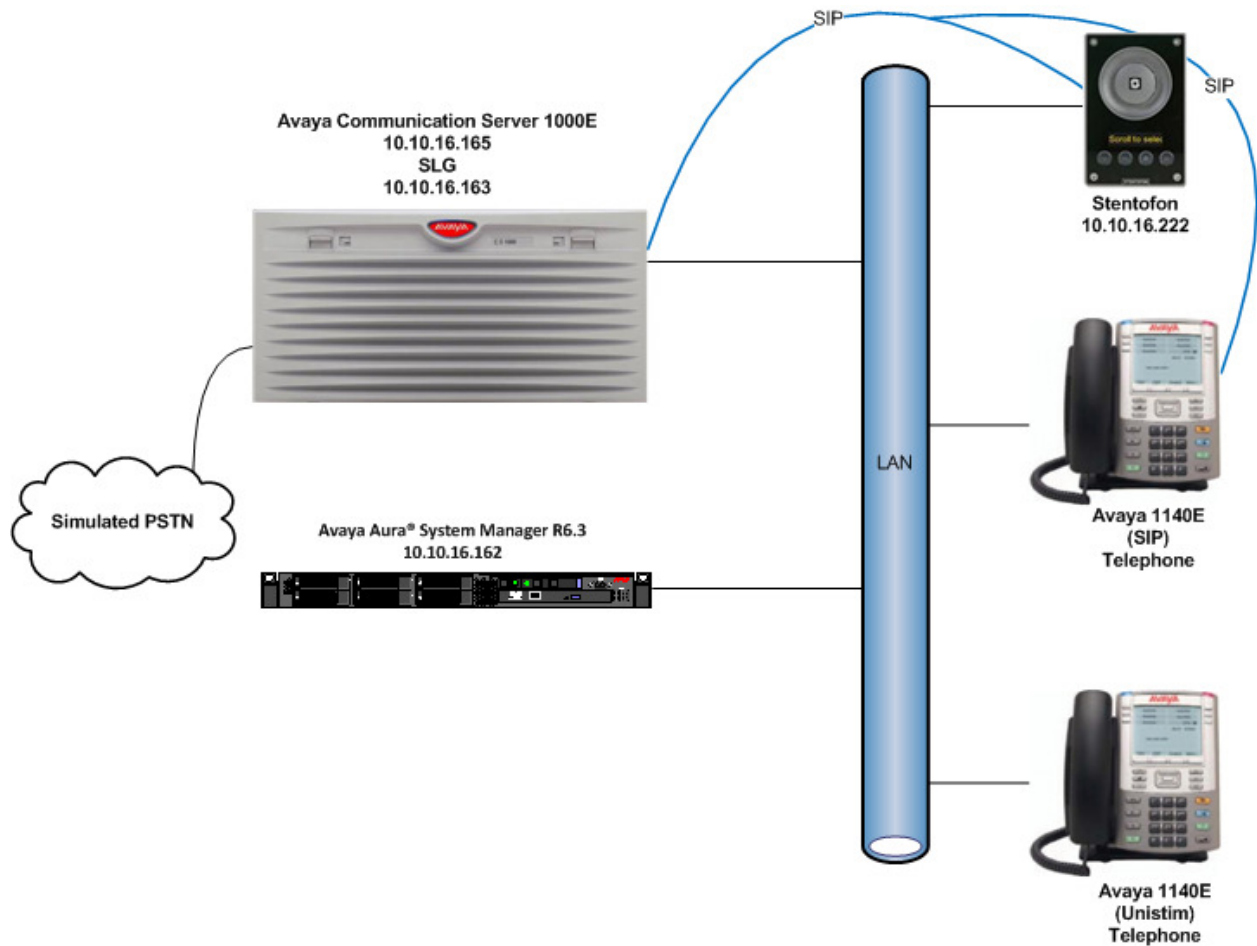


Figure 1: Avaya Communication Server 1000E and Zenitel Stentofon Configuration

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 be 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**.

AVAYA Avaya Aura® System Manager 6.3 Help | Logout

Host Name: 10.10.16.162 User Name: admin

Elements

New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Search Reset

Add... Edit... Delete

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
1 <input type="checkbox"/>	smgr63.devconnect.local (primary)	Base OS	7.6	10.10.16.162	Base OS element.
2 <input type="checkbox"/>	EM on cs1krp	CS1000	7.6	10.1.16.164	New element.

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.

The screenshot displays the Avaya CS1000 Element Manager interface. The left sidebar contains a navigation menu with the following items: UCM Network Services, Home, Links, Virtual Terminals, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, Nodes Servers Media Cards (highlighted with a red box), Maintenance and Reports, Media Gateways, Zones, Host and Route Tables, Network Address Translation, and QoS Thresholds. The main content area shows the 'Node Details (ID: 163 - SIP Line, LTPS, Gateway (SIPGw))' page. The page header indicates the managing IP is 10.1.16.164 and the user is admin. The breadcrumb trail is System » IP Network » IP Telephony Nodes » Node Details. The form contains the following fields: Node ID (163), Call server IP address (10.1.16.164), Embedded LAN (ELAN) Gateway IP address (10.1.16.1), Subnet mask (255.255.255.0), TLAN address type (IPv4 only selected), and Telephony LAN (TLAN) Node IPv4 address (10.10.16.163, highlighted with a red box). The Subnet mask for the TLAN is 255.255.255.0.

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.

The screenshot displays the Avaya CS1000 Element Manager interface. The left sidebar contains a navigation menu with categories like UCM Network Services, Home, Links, Virtual Terminals, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, and Nodes. The main content area shows the configuration for Node ID 163, specifically the SIP Line Configuration Details. The 'General' tab is active, showing the 'SIP Line Gateway Application' section with a checked box for 'Enable gateway service on this node'. Below this, the 'General' section contains three input fields: 'SIP domain name' (value: devconnect.local), 'SLG endpoint name' (value: CS1KRP), and 'SLG Local Sip port' (value: 5070). The 'Virtual Trunk Network Health Monitor' section is also visible, with a checkbox for 'Monitor IP addresses (listed below)' and a 'Monitor IP' input field with an 'Add' button.

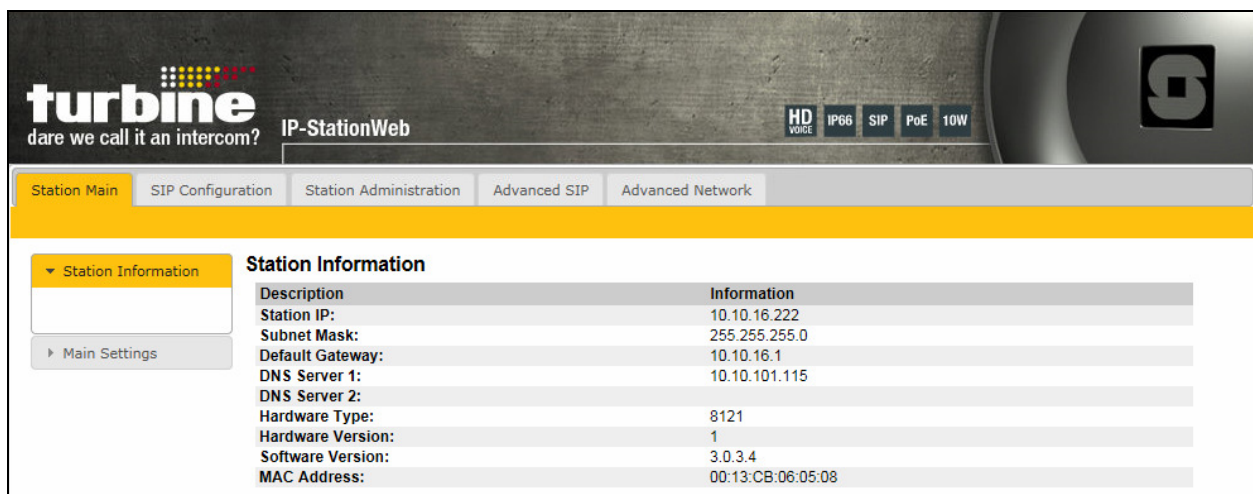
6. Configure Zenitel Stentofon Turbine Intercom

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**



The screenshot displays the Stentofon Turbine web interface. At the top left is the 'turbine' logo with the tagline 'dare we call it an intercom?'. To the right of the logo is the text 'IP-StationWeb'. Further right are several feature icons: 'HD VOICE', 'IP66', 'SIP', 'PoE', and '10W'. Below the header is a navigation bar with tabs: 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The 'Station Main' tab is active. On the left side, there is a sidebar with a 'Station Information' dropdown menu and a 'Main Settings' button. The main content area is titled 'Station Information' and contains a table with the following data:

Description	Information
Station IP:	10.10.16.222
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.101.115
DNS Server 2:	
Hardware Type:	8121
Hardware Version:	1
Software Version:	3.0.3.4
MAC Address:	00:13:CB:06:05:08

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.

The screenshot shows a web interface for configuring SIP settings. At the top, there are navigation tabs: Station Main, SIP Configuration, Station Administration, Advanced SIP, and Advanced Network. On the left, a sidebar menu has 'Station Information' and 'Main Settings' (highlighted). The main content area is titled 'Station Mode' and contains four radio button options: 'Use SIP' (selected), 'Use Alphacom', 'Use Pulse', and 'Use Pulse Server'. Below this is a 'Turbine Frontboard' section with a dropdown menu set to 'Scrolling Station (TCIS-6)'. The 'IP Settings' section has 'DHCP' selected and 'Static IP' unselected. Below are input fields for IP address, subnet mask, gateway, and DNS servers, along with a hostname field and a checked 'Read IP Address' checkbox.

IP-address:	10	-	10	-	16	-	222
Subnet-mask:	255	-	255	-	255	-	0
Gateway:	10	-	10	-	16	-	1
DNS Server 1:	10	-	10	-	101	-	115
DNS Server 2:	0	-	0	-	0	-	0

Hostname: zenite1060508

Read IP Address:

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**.

Station Main		SIP Configuration	Station Administration	Advanced SIP	Advanced Network																										
<div style="display: flex;"> <div style="border: 1px solid gray; padding: 5px; width: 20%;"> <p>▼ SIP Settings</p> <p>▶ Audio Settings</p> <p>▶ Direct Access Key Settings</p> <p>▶ Relay Settings</p> <p>▶ Time Settings</p> <p>▶ Address Book</p> <p>▶ I/O Settings</p> </div> <div style="margin-left: 10px;"> <h3>Account Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td>Zenitel3020</td> </tr> <tr> <td>Directory Number (SIP ID):</td> <td>3020</td> </tr> <tr> <td>Server Domain (SIP):</td> <td>devconnect.local</td> </tr> <tr> <td>Backup Domain (SIP):</td> <td></td> </tr> <tr> <td>Backup Domain 2 (SIP):</td> <td></td> </tr> <tr> <td>Registration Method:</td> <td>Parallel</td> </tr> <tr> <td>Authentication User Name:</td> <td>3020</td> </tr> <tr> <td>Authentication Password:</td> <td>••••</td> </tr> <tr> <td>Register Interval:</td> <td>600 (min. 60 seconds)</td> </tr> <tr> <td>Outbound Proxy [optional]:</td> <td>cs1krp.devconnect.local Port: 5070</td> </tr> <tr> <td>Outbound Backup Proxy [optional]:</td> <td>Port: 5060</td> </tr> <tr> <td>Outbound Backup Proxy 2 [optional]:</td> <td>Port: 5060</td> </tr> </tbody> </table> </div> </div>						Description	Configuration	Display Name:	Zenitel3020	Directory Number (SIP ID):	3020	Server Domain (SIP):	devconnect.local	Backup Domain (SIP):		Backup Domain 2 (SIP):		Registration Method:	Parallel	Authentication User Name:	3020	Authentication Password:	••••	Register Interval:	600 (min. 60 seconds)	Outbound Proxy [optional]:	cs1krp.devconnect.local Port: 5070	Outbound Backup Proxy [optional]:	Port: 5060	Outbound Backup Proxy 2 [optional]:	Port: 5060
Description	Configuration																														
Display Name:	Zenitel3020																														
Directory Number (SIP ID):	3020																														
Server Domain (SIP):	devconnect.local																														
Backup Domain (SIP):																															
Backup Domain 2 (SIP):																															
Registration Method:	Parallel																														
Authentication User Name:	3020																														
Authentication Password:	••••																														
Register Interval:	600 (min. 60 seconds)																														
Outbound Proxy [optional]:	cs1krp.devconnect.local Port: 5070																														
Outbound Backup Proxy [optional]:	Port: 5060																														
Outbound Backup Proxy 2 [optional]:	Port: 5060																														

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

Call Settings

Description	Configuration
Enable Auto Answer:	<input checked="" type="checkbox"/>
Auto Answer Delay:	<input type="text" value="0"/> seconds. Max 30 seconds.
Delay Call Setup:	<input type="text" value="0"/> seconds. Max 60 seconds. Delays call setup using DAK/Input buttons.
Overlap dialing:	<input type="checkbox"/>
DTMF method:	SIP INFO <input type="button" value="v"/>
RTP Timeout value:	<input type="text" value="0"/> seconds. 0 = RTP Timeout Disabled.
Codec g722:	High Priority <input type="button" value="v"/>
Codec g711a:	Low Priority <input type="button" value="v"/>
Codec g711u:	Medium Priority <input type="button" value="v"/>

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

Station Main
SIP Configuration
Station Administration
Advanced SIP
Advanced Network

- ▶ SIP Settings
- ▶ Audio Settings
- ▶ Direct Access Key Settings
- ▼ Relay Settings
- ▶ Time Settings
- ▶ Address Book
- ▶ I/O Settings

Relay Settings

Choose Relay To Configure: Relay 1

Relay 1 Settings

Description	Configuration
Remote Digit For Relay On:	<input type="button" value="v"/> 0
Remote Digit For Relay Off:	<input type="button" value="v"/> 1
Remote Digit For Relay Slow Flash :	<input type="button" value="v"/> 2
Remote Digit For Relay Fast Flash:	<input type="button" value="v"/> 4
Remote Digit For Relay Toggle:	<input type="button" value="v"/> 5
Remote Digit For Timed Relay On:	<input type="button" value="v"/> 6
Timed Relay Duration:	<input type="text" value="3"/> seconds.
Outgoing Ringing:	<input type="button" value="v"/> -
Incoming Ringing:	<input type="button" value="v"/> -
Outgoing Call:	<input type="button" value="v"/> -
Incoming Call:	<input type="button" value="v"/> -
Group Call (Pulse mode only):	<input type="button" value="v"/> -
Idle:	<input type="button" value="v"/> -
Error (Not Registered):	<input type="button" value="v"/> -

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.

The screenshot shows the 'Address Book' configuration page. The left sidebar contains navigation options: SIP Settings, Audio Settings, Direct Access Key Settings, Relay Settings, Time Settings, and Address Book (selected). The main content area is titled 'Address Book' and contains the following settings:

Description	Configuration	
Default Display Text:	Scroll to Select	
Font Size	12	
OLED Brightness	Default	Higher brightness reduces OLED lifetime
Start Scrolling After:	5 minutes	0 is off. Scrolling increases OLED lifetime
Upload Address Book:	<input type="text"/>	<input type="button" value="Browse..."/> Must be .csv file with format: number;text

Below the settings is a button. At the bottom, a note reads: 'Note: Using new address book requires application reboot'.

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**.

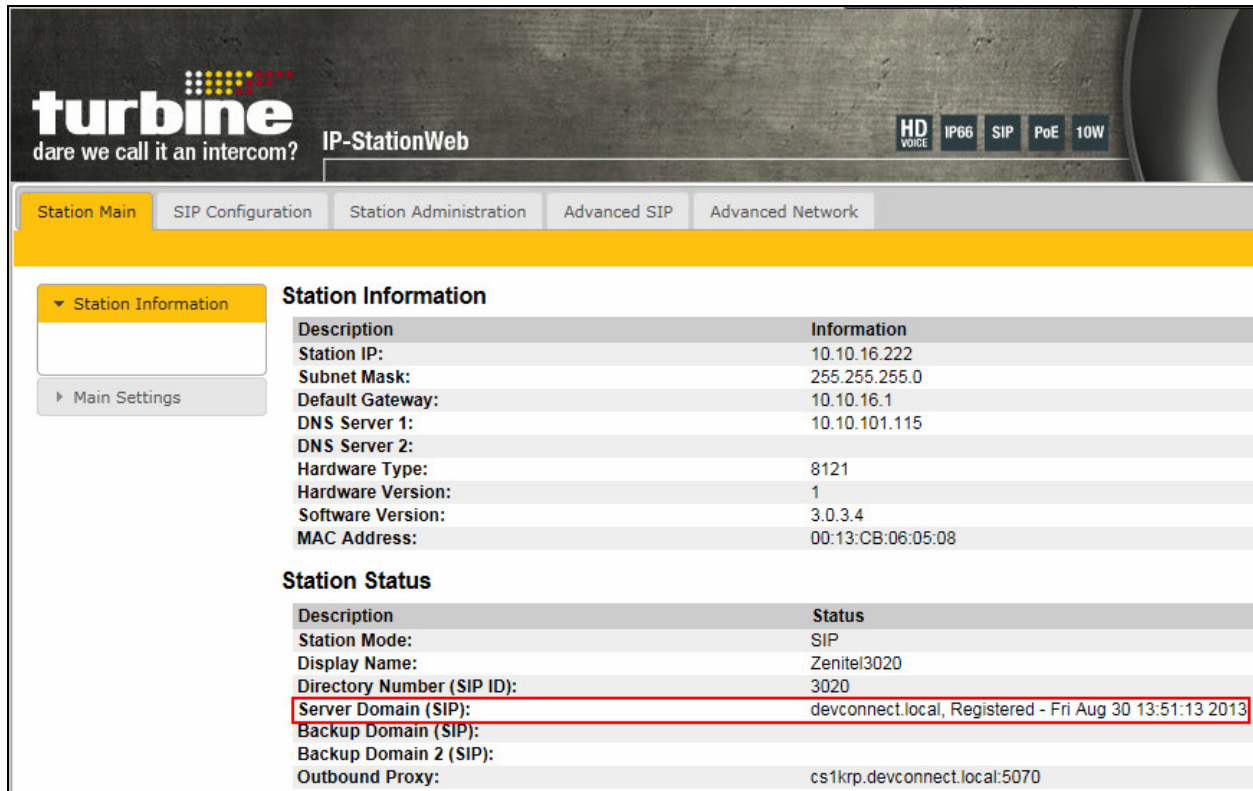
Example Below shows that the Stentofon Turbine 3020 is registered.

```
[admin@cslkrp ~]$ 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 Stentofon 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.



The screenshot displays the Stentofon Turbine web interface. The top navigation bar includes 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The 'Station Information' section is active, showing a table of station details. Below it, the 'Station Status' section shows the registration status of the SIP server domain.

Description	Information
Station IP:	10.10.16.222
Subnet Mask:	255.255.255.0
Default Gateway:	10.10.16.1
DNS Server 1:	10.10.101.115
DNS Server 2:	
Hardware Type:	8121
Hardware Version:	1
Software Version:	3.0.3.4
MAC Address:	00:13:CB:06:05:08

Description	Status
Station Mode:	SIP
Display Name:	Zenitel3020
Directory Number (SIP ID):	3020
Server Domain (SIP):	devconnect.local, Registered - Fri Aug 30 13:51:13 2013
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Outbound Proxy:	cs1krp.devconnect.local:5070

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>.

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