Quick Installation & Configuration Guide

SDS-1

SIP Desk Station with Display & Dual LAN

The SDS-1 is a product developed for Zenitel Norway AS and is primarily used as part of a Zenitel Norway AS IP Intercom solution. The SDS-1 is not pre-configured to support or carry emergency calls to any type of hospital, law enforcement agency, medical care unit ("Emergency Service(s)") or any other kind of Emergency Service. You must make additional arrangements to access Emergency Services. It is your responsibility to purchase SIP-compliant Internet telephone service, properly configure the SDS-1 to use that service, and periodically test your configuration to confirm that it works as you expect. If you do not do so, it is your responsibility to purchase traditional wireless or landline telephone services to access Emergency Services.

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Precautions

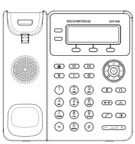
WARNING: Please DO NOT power cycle the SDS-1 during system boot-up or firmware upgrade. You may corrupt firmware images and cause the unit to malfunction.

Overview

- SDS-1 SIP Desk Station with Display and Dual LAN (Item Number: 1490000010)

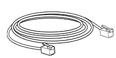
SDS-1 is a Small Business HD IP phone that features 2 lines with 2 SIP accounts, 132x48 backlit graphical LCD, 3 XML programmable context-sensitive softkeys, dual network ports with PoE and 3-way conference. The SDS-1 delivers HD wideband audio, superb full-duplex hands-free speakerphone with advanced acoustic echo cancellation, advanced security protection for privacy, leading edge SIP intercom features and integration with the Pulse and AlphaCom IP Intercom platforms developed by Zenitel Norway AS.

Package Contents



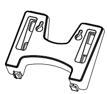
1 x Phone Main Case

1 x Handset

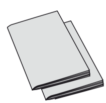


1 x Ethernet Cable

1 x Phone Cord



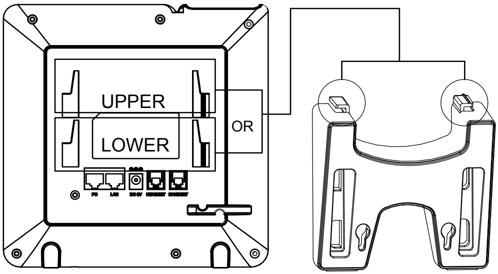
1 x Phone Stand



1 x Quick Installation Guide
1 x GPL License

Phone Installation

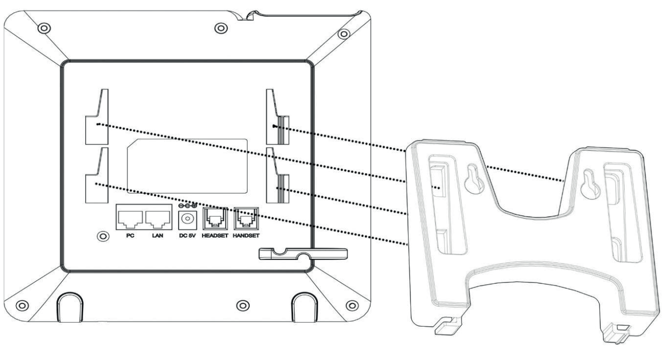
Installing the Phone with Phone Stand



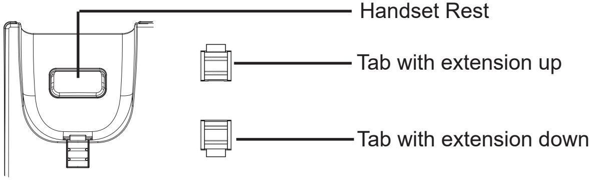
1. Insert the hooks on top of the stand into the slots on the back of the phone
    - You can either use the upper OR lower slots
2. Firmly slide the stand upward to lock it in place

Installing the Phone with Wall Mount

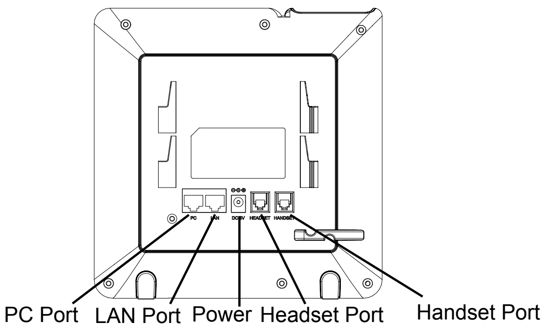
1. Insert all 4 hooks located at the front of the wallmount into the slots on the back of the phone.
2. Firmly slide the wall mount upward to lock it in place.
3. Attach the phone to the wall via the wall mount holes.



4. Pull out the tab from the handset cradle (see figure below).
5. Rotate the tab and plug it back into the slot with the extension up to hold the handset while the phone is mounted on the wall.



Connecting the Phone

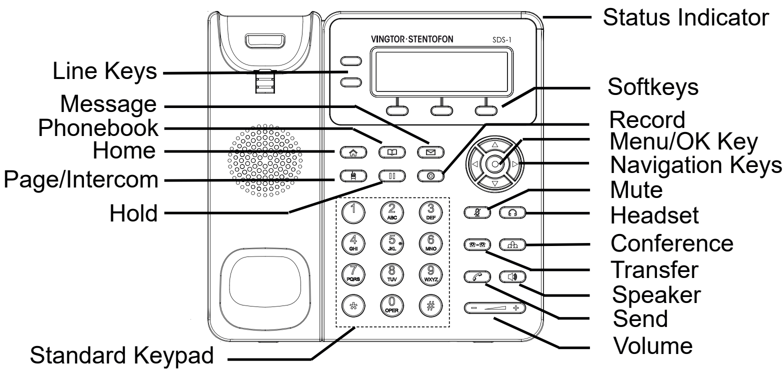


1. Connect the handset and main phone case with the phone cord.

2. Connect the LAN port of the phone to the RJ-45 socket of a PoE switch using the Ethernet cable.
    - The LCD will display provisioning or firmware upgrading information. Before continuing, please wait for the date/time display to appear.

3. Using the web configuration interface or the keypad configuration menu, you can further configure the phone using either a static IP or DHCP.

Tips for Using the Keypad



1. To access the menu, press the round **Menu** key.
2. Navigate the menu by using the **Up/Down** and **Left/Right** arrow-keys.
3. Press the round **Menu** key to confirm a menu selection.
4. The phone automatically exits menu mode when there is an incoming call, the phone goes off-hook, or when menu mode is left idle for 60 seconds.

Phone Configuration

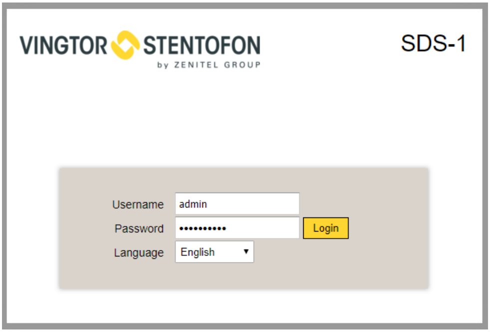
For further information on the configuration of AlphaCom, SIP, and Pulse, please go to wiki.zenitel.com.

Configuring the SDS-1 Using the Keypad

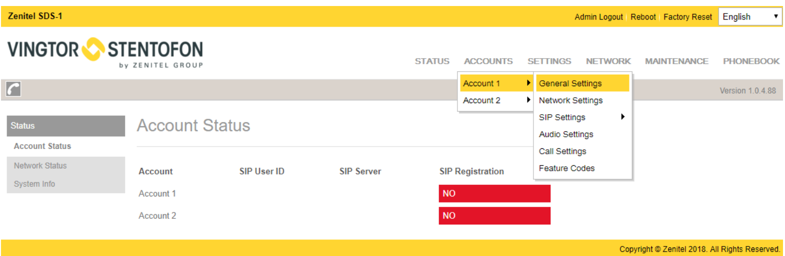
1. Make sure the phone is idle.
2. Press the **Menu** key to access the keypad menu to configure the phone.
3. Select **Phone** > **SIP** > **Account** to configure settings for **SIP Server** (AlphaCom or Pulse), **SIP User ID** (Extension Number), SIP Auth ID and SIP Password.
4. Follow the menu options to configure the basic features of the phone, e.g. the IP address if using a static IP.

Configuring the SDS-1 Using Web Browser

1. Ensure your phone is properly powered up and connected to the Internet.
2. Press the **Menu** key to enter the menu of the phone.
3. Navigate to **Status** > **Network Status** and press the **Menu** key to check the IP address.
4. Enter the phone's IP address in your PC's browser.



5. Log in by entering the default **Username**: **admin** and **Password**: **alphaadmin**

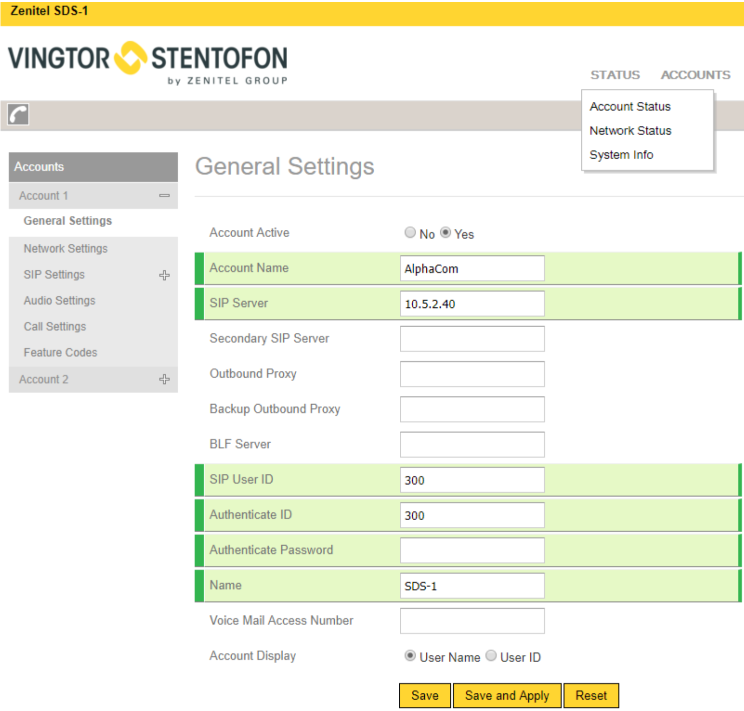


6. Register the account on the SDS-1 by selecting **ACCOUNTS** > **Account 1/2** > **General Settings** to configure **Account Name**, **SIP Server** (AlphaCom or Pulse), **SIP User ID** (Extension Number).

AlphaCom Configuration

AlphaCom SDS-1 Account Setup

- Select **ACCOUNTS** > **Account 1** > **General Settings**



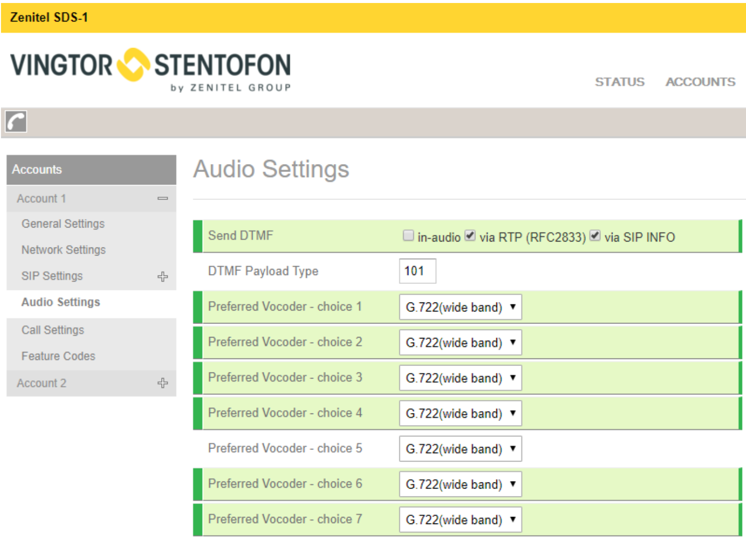
- Enter the values shown above for the parameters

Account Active: Check **Yes** button
SIP Server: IP address of **AlphaCom** server
SIP User ID: Directory Number of **SDS-1** phone
Authenticate ID: Same as **SIP User ID**

Note: *For changes to take effect, it may be necessary to temporarily disable the account. First check the* **No** *button for* **Account Active**, *then click* **Save and Apply**. *Once this is done, re-enable the account by checking the* **Yes** *button for* **Account Active** *followed by* **Save and Apply** *again.*

AlphaCom SDS-1 Audio Settings

- Check in AlphaPro under **Users & Stations** the codec that has been selected for the SIP phone (normally **G722**)
- Select **Account 1** > **Audio Settings**

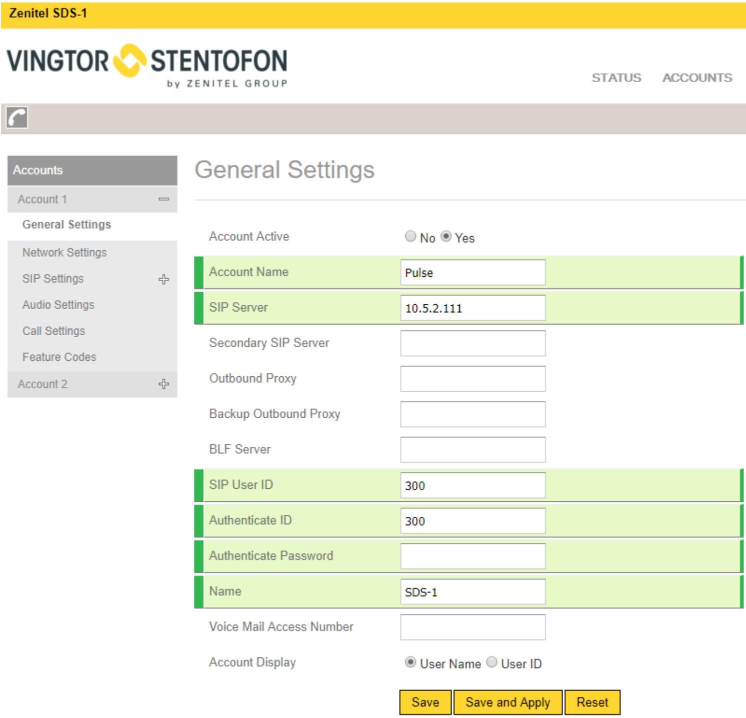


- Set all codecs from the **Preferred Vocoder** list to the one defined in AlphaPro, i.e. **G722**.

Pulse Configuration

Pulse SDS-1 Account Setup

- Select **ACCOUNTS** > **Account 1** > **General Settings**



- Enter the values shown above for the parameters

Account Active: Check **Yes** button

SIP Server: IP address of intercom station set as **Pulse Server**

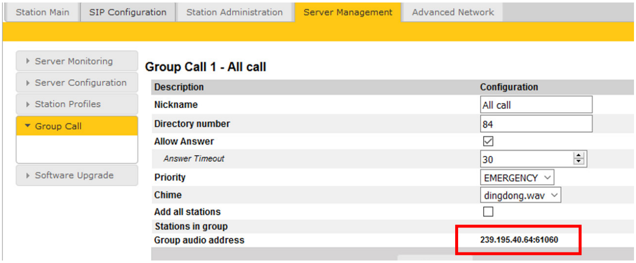
SIP User ID: Directory Number of the **SDS-1** phone

Authenticate ID: Same as **SIP User ID**

Pulse Group Call

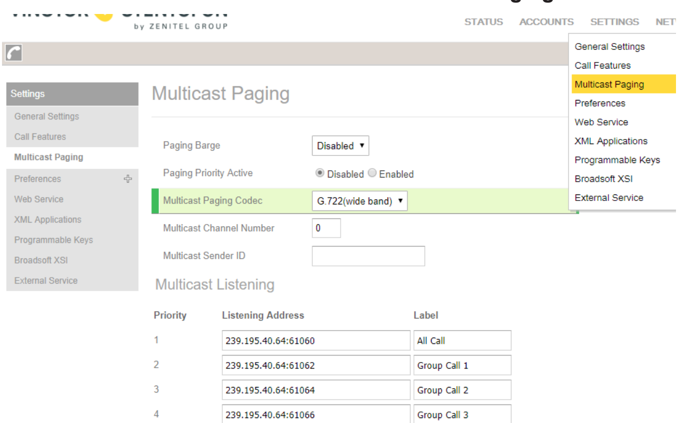
The Pulse Server transmits group calls using IP multicast paging. Each group call uses its own unique multicast IP address. To find the multicast IP address:

- Log into the Pulse Server
- Select **Server Management** > **Group Call**



Make a note of the multicast addresses under **Group audio address**.

- Log into the SDS-1
- Select **SETTINGS** > **Multicast Paging**



- Set **Multicast Paging Codec** to **G722**
- Enter the multicast addresses under **Listening Address**
- Click **Save and Apply** and **Reset**

When a Group Call is activated, the SDS-1 will automatically broadcast the audio in the loudspeaker. The SDS-1 will display the text of the Group Call as entered under **Nickname**. If the SDS-1 is busy in a regular call when a Group Call is made, it will by default not play the Group Call audio. If **Paging Barge** is set to value **2** or higher, the current call will be placed On Hold, and the Group Call audio will be broadcast. When the Group Call is ended, press the Hold button to resume the regular call.

VINGTOR STENTOFON

Zenitel Norway AS
Sandakerveien 24C
0403 Oslo, Norway

WE HAVE 24/7 CUSTOMER SERVICE

GLOBAL: +47 4000 2700

USA: 800 654 3140

Mon-Fri 09:00 to 17:00 CST

cs@zenitel.com

For Warranty and RMA information, please visit www.zenitel.com

DOC NUMBER
A100K11892 6.11.18

customer.service@zenitel.com