



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 Phone Cord

1 x Phone Stand



1 x Ethernet Cable



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.
- 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.
- 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.

	SDS-1		
Username Password Language	admin ••••••• English •	Login	

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

Zenitel SDS-1							Admin Logout	Reboot Factory Reset	English 🔹
	ENTOFON			STATUS	ACCOUNTS	SETTINGS	NETWORK	MAINTENANCE	PHONEBOOK
٢					Account 1	General S	Settings		Version 1.0.4.88
Status Account Status Network Status System Info	Account S Account 1 Account 2	Status SIP User ID	SIP Server	SIP NO	Account 2	Network 5 SIP Settir Audio Set Call Settir Feature C	Settings Igs + tings Ings codes		
							Co	pyright © Zenitel 2018. A	Il Rights Reserved.

 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

Zenitel SDS-1			
	ENTOFON ZENITEL GROUP		STATUS ACCOUNTS
<u>ſ</u>			Account Status
Accounts Account 1	General Settings		Network Status System Info
General Settings	Account Active	○ No ● Yes	
Network Settings	Account Name	AlphaCom	
Audio Settings	SIP Server	10.5.2.40	
Call Settings	Secondary SIP Server		
Account 2	Outbound Proxy		
	Backup Outbound Proxy		
	BLF Server		
	SIP User ID	300	
	Authenticate ID	300	
	Authenticate Password		
	Name	SDS-1	
	Voice Mail Access Number		
	Account Display	● User Name ^O User ID	
		Save Save and Apply	/ Reset

• 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

Zenitel SDS-1							
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<u>ſ</u>							
Accounts Account 1		Audio Settings					
General Settings		Send DTMF	🗆 in-audio 🗹 via RTP (RFC2833) 🗹 via SIP INFO				
SIP Settings	÷	DTMF Payload Type	101				
Audio Settings		Preferred Vocoder - choice 1	G.722(wide band)				
Call Settings Feature Codes		Preferred Vocoder - choice 2	G.722(wide band) •				
Account 2	ella	Preferred Vocoder - choice 3	G.722(wide band) ▼				
		Preferred Vocoder - choice 4	G.722(wide band)				
		Preferred Vocoder - choice 5	G.722(wide band) ¥				
		Preferred Vocoder - choice 6	G.722(wide band)				
		Preferred Vocoder - choice 7	G.722(wide band)				

• 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

Zenitel SDS-1					
	STE	NTOFON ENITEL GROUP		STATUS	ACCOUNTS
~					
Accounts		General Settings			
General Settings		Account Active	◯ No ම Yes		
SIP Settings	÷	Account Name	Pulse		
Audio Settings		SIP Server	10.5.2.111		
Call Settings		Secondary SIP Server			
Feature Codes	4	Outbound Proxy			
		Backup Outbound Proxy			
		BLF Server			
		SIP User ID	300		
		Authenticate ID	300		
		Authenticate Password			
		Name	SDS-1		
		Voice Mail Access Number			
		Account Display	◉ User Name ◯ User ID		
			Save Save and Appl	y Reset	

• 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 •

ation Main	SIP Config	uration	Station Administration	Server Management	Advanced Networ	rk
 Server Moni 	itoring					
· Server Hom	iconing	Group	o Call 1 - All call			
 Server Conf 	figuration	Desc	ription		C	onfiguration
Station Prof	iles	Nickr	iame		A	ll call
 Group Call 		Direc	tory number		8	14
		Allow	Answer		6	2
		Ans	wer Timeout		3	0
Software Up	ograde	Priori	ity		E	EMERGENCY ~
		Chim	e		c	dingdong.wav 🖂
		Add a	III stations		0	
		Statio	ons in group			
		Group	o audio address		23	39.195.40.64:61060

Make a note of the multicast addresses under Group audio address. Log into the SDS-1

NITEL GROUP	aging		STATUS	ACCOUNT	S SETTINGS General Settings Call Features Multicast Paging Preferences
Iulticast Pa Paging Barge	aging				General Settings Call Features Multicast Paging Preferences
Iulticast Pa Paging Barge	aging				Call Features Multicast Paging Preferences
Iulticast Pa	aging				Multicast Paging Preferences
Paging Barge	aging				Preferences
Paging Barge					
Paging Barge					Web Service
		Disabled •			XML Applications
Paging Priority Activ	e	Disabled	d		Programmable Keys
					External Service
Multicast Paging Co	dec	G.722(wide band) •			External Service
Multicast Channel N	umber	0			
Multicast Sender ID					
Iulticast Liste	ning				
iority Lister	ning Address		Label		
239.1	195.40.64:6106	0	All Call		
239.1	195.40.64:6106	2	Group Call 1		
239.1	195.40.64:6106	4	Group Call 2		
239.1	195.40.64:6106	6	Group Call 3		
	Multicast Paging Co Multicast Channel N Multicast Sender ID Iulticast Liste Iority Lister 239.1 239.1 239.1 239.1 239.1 239.1 239.1 239.1	Multicast Paging Codec Multicast Channel Number Multicast Sender ID Iulticast Listening 239.195.40.64:6106 239.195.40.64:6106 239.195.40.64:6106 239.195.40.64:6106 Set Multicast	Authorset G 722(wide band) Multicast Sender ID 0 Multicast Sender ID 0 Julticast Listening 0 239.195.40.64/s1060 239.195.40.64/s1062 239.195.40.64/s1064 239.195.40.64/s1064 239.195.40.64/s1066 0 Set Multicast Paging Color	Multicast Paging Codec G 722(vide band) • Multicast Channel Number 0 Multicast Sender ID Iulticast Listening 239.195.40.64:61060 All Call 239.195.40.64:61062 Group Call 1 239.195.40.64:61064 Group Call 2 239.195.40.64:61064 Group Call 3 Set Multicast Paging Codec to G	Multicast Paging Codec G 722(wide band) Multicast Channel Number

- 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.



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