Administering Avaya E129 SIP Deskphone

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- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
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FCC/Industry Canada Radiation Exposure Statement

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US/Canada

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Australia Warning

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EU Countries

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DenAn Power Cord Regulatory Notice for Japan

⚠ Warning

Please be careful of the following while installing the equipment:

- Please only use the Connecting cables, power cord, AC adaptors shipped with the equipment or specified by Avaya to be used with the equipment. If you use any other equipment, it may cause "failures, malfunctioning or fire".
- Power cords shipped with this equipment must not be used with any other equipment. In case the above guidelines are not followed, it may lead to death or severe injury

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Power over Ethernet (PoE) warning

This equipment must be connected to PoE networks without routing to the outside plant.

Contents

Chapter 1: Introduction

Purpose

This document provides deskphone administration and maintenance procedures that you can perform through different interfaces.

Intended audience

This document is for administrators, implementation engineers, and BusinessPartners responsible for deskphone administration and maintenance.

Document changes since last issue

The following changes have been made to this document since the last issue:

- Removed references to UDP and NAT.

Related resources

Documentation

Document number	Title	Use this document to:	Audience
Implementing			
16–604370	Installing and Maintaining Avaya E129 SIP Deskphone	See the install and upgrade procedures for the Avaya E129 SIP deskphone.	Administrators and network engineers

Table continues…

Document number	Title	Use this document to:	Audience
Using			
16–604368	Using Avaya E129 SIP Deskphone	See the capabilities of Avaya E129 SIP deskphone and to learn about how various features work.	Users and administrators
16–604373	Avaya E129 SIP Deskphone Quick Reference	See frequently used tasks.	Users and administrators

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Chapter 2: Getting started

Administration features

You can administer the following features of the deskphone:

- SIP settings
- Network settings
- Phone settings, such as ring tone, time display format, and phone screen contrast
- Security settings

You can administer the features:

- Locally through a deskphone
- Remotely through a Web GUI of the respective phone
- Centrally through the configuration file

Connection ports

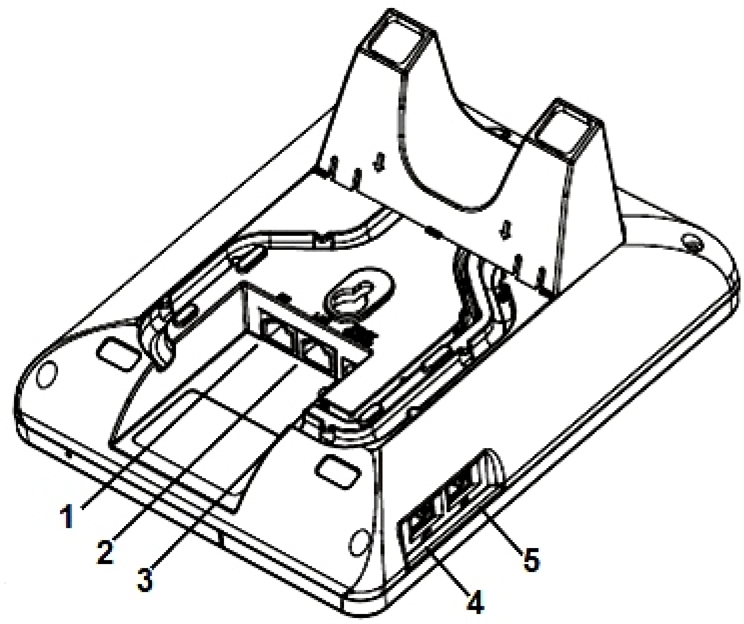

Figure 1: Connection ports at the back of the deskphone

Number	Port name	Description
1	PC	A 10/100–Mbps RJ45 port to connect to a personal computer or a mobile computer
2	LAN	A 10/100–Mbps RJ45 port to connect to PoE-supported Ethernet
3	Power	A 5–V DC port to connect to the power supply
4	Handset	An RJ9 port to connect the handset
5	Headset	An EHS-suported RJ9 port to connect to a Plantronics headset

Button layout

Number	Button name	Description
1	Softkeys	Display screen-specific commands.
2	Phonebook	Displays the contacts list.
3	Transfer	Transfers the call to the selected number.
4	Mute	Mutes and unmutes the microphone.
5	Speaker	Activates and deactivates the speakerphone.
6	Volume	Increases and decreases the volume.
7	Call	Dials the number that you enter.
8	Flash	Puts the current call on hold and brings up the second line for dialing or answering the second call.
9	Conference	Starts the conference.
10	Message	Dials the voice mail server.

Table continues…

Number	Button name	Description
11	Up, Down, Right, and Left navigation keys	Navigates between various menu options. You can use the Left navigation key to perform the back function that takes you one level up the menu options.
12	Menu	Displays the menu or selects the highlighted option.
13	Message Waiting Indicator	Turns red if you get a voice mail.

Chapter 3: Administering through the deskphone

You can administer the deskphone locally through the administration options that the deskphone provides. You can administer features such as the SIP account, network settings, and contacts.

You might need to enter the password in case you configured the password through the Web GUI or through the configuration file.

Note:

You can use the keypad to input letters and special symbols. To insert a period, press ***** multiple times to cycle to the period symbol. To insert a colon, press **#** .

Related links

Configuring the SIP account settings on page 14
SIP account field descriptions on page 15
Configuring the upgrade phone settings on page 15
Upgrade phone field descriptions on page 16
Resetting the phone to factory default on page 16
Configuring Layer2 QoS settings on page 17
Layer2 QoS field descriptions on page 17
Configuring the network settings on page 17
Network field descriptions on page 18

Configuring the SIP account settings

About this task

Use the following procedure to configure SIP account parameters of the deskphone.

Note:

If you enabled the Constraint Mode option, the deskphone does not display the **Config** menu option. In this case, use the **Admin Login** menu option to view all hidden menu options.

Procedure

1. Press **Menu** > **Config** > **SIP**.

2. Enter the required values for each SIP parameter.
3. Select **Save**, and press **Menu**.

Related links

Administering through the deskphone on page 14

SIP account field descriptions

Name	Description
SIP Proxy	The IP address or the URL of the SIP proxy server. The supported syntax is `IP/Host:port`.
SIP Transport	The SIP transport protocol. The options are: • **TCP** • **TLS/TCP**
Audio	The audio codec that the deskphone uses. The options are: • **PCMU** • **PCMA** • **G.722** • **G.723.1** • **G.726–32** • **G.729(A/B)** • **iLBC**

Related links

Administering through the deskphone on page 14

Configuring the upgrade phone settings

About this task

Use the following procedure to configure settings to upgrade the phone. Ensure that you upgrade the deskphone during the off hours.

Procedure

1. Press **Menu** > **Config** > **Upgrade**.
2. Enter the required values for each upgrade parameter.

3. Select **Save**.
4. Press the **Yes** softkey when the deskphone displays the prompt to reboot.

Related links

Administering through the deskphone on page 14

Upgrade phone field descriptions

Name	Description
Firmware Server	The URL or the IP address of the server from where the deskphone downloads the latest firmware.
Config Server	The URL or the IP address of the server from where the deskphone downloads the configuration file.
Upgrade Via	The protocol that the deskphone uses to download the firmware. You can select from the following options: • **TFTP** • **HTTP** • **HTTPS**

Related links

Administering through the deskphone on page 14

Resetting the phone to factory default

About this task

Use the following procedure to reset SIP and network parameters to the default factory settings.

Procedure

1. Press **Menu** > **Config** > **Factory Reset**.
2. Press **OK** to confirm the resetting.

   The phone resets all SIP and network parameters to the default factory settings.

Related links

Administering through the deskphone on page 14

Configuring Layer2 QoS settings

About this task

Use the following procedure to set the VLAN parameters.

Procedure

1. Press **Menu** > **Config** > **Layer2 QoS**.
2. Enter the required values for each VALN parameter.

Related links

Administering through the deskphone on page 14

Layer2 QoS field descriptions

Name	Description
802.1Q/VLAN Tag	The VLAN tag for the Layer2 QoS packets.
priority value	The priority value for the Layer2 QoS packets.
Reset Vlan Config	The option to reset the VLAN tag value.

Related links

Administering through the deskphone on page 14

Configuring the network settings

About this task

Use the following procedure to configure network parameters, such as the IP mode and the Gateway IP address.

Procedure

1. Press **Menu** > **Network Config**.
2. Enter the required value for each network parameter.
3. Select **Back**, and press **Menu**.
4. To immediately bring into effect the updated values, press **Reboot** , else press **No**.

Related links

Administering through the deskphone on page 14

Network field descriptions

Name	Description
IP Setting	Specifies the IP mode that the deskphone uses for the IP address. Select from the following options: • **DHCP** • **Static IP** • **PPPoE** **Note:** Avaya E129 SIP Deskphone does not support PPPoE.
PPPoE Settings	Avaya E129 SIP Deskphone does not support PPPoE.
IP	Specifies the IP address of the deskphone when the IP mode is static.
Netmask	Specifies the subnet mask that the deskphone uses when the IP mode is static.
Gateway	Specifies the gateway that the deskphone uses when the IP mode is static.
DNS Server 1	Specifies the default DNS server that the deskphone uses when the IP mode is static.
DNS Server 2	Specifies the second DNS server that the deskphone uses when the IP mode is static.
802.1X	Specifies the 802.1X mode details.

802.1X field descriptions

Name	Description
Mode	Specifies the option to enable or disable the 802.1X mode. You can select from the following options: • **Disable** • **EAP-MD5**
Identity	Specifies the identity of the 802.1X EAP-MD5 mode.
MD5 Password	Specifies the MD5 password for the 802.1X EAP-MD5 mode.

Related links

Administering through the deskphone on page 14

Chapter 4: Administering through the Web GUI

You can administer the deskphone remotely through a Web GUI that Avaya provides with deskphones. You can administer features such as the SIP account, network settings, and contacts.

You require a password to gain access to the administrative options available on the Web GUI.

Related links

Gaining access to the Web GUI on page 19
Configuring the accounts audio settings on page 28
Accounts audio settings field descriptions on page 28
Configuring the accounts call settings on page 30
Accounts call settings field descriptions on page 30

Gaining access to the Web GUI

Before you begin

You must get the following:

- The administrator password
- The IP address of the phone that you want to administer

About this task

Use the following procedure to gain access to the Web GUI of the respective phone that you want to administer. The Web GUI supports only the English language.

Procedure

1. Start a browser.
2. In the address field of the browser, type the IP address of the phone, and press `Enter`.

   The system displays the Login page.
3. In the **Password** field, enter the administrator password.
4. In the **Language** field, click the required language.
5. Click **Login**, or press `Enter`.

The system displays the Account Status page of the Web GUI.

Related links

Administering through the Web GUI on page 19

Managing the account settings

Configuring the accounts general settings

About this task

You must restore the deskphone to the factory default if you are configuring the deskphone for another profile.

Procedure

1. Click **Accounts** > **Account 1** > **General Settings**.

   The system displays the General Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Accounts general settings field description

Name	Description
Account Active	Specifies the status that indicates whether the account is active. You can select from the following options: • **No** • **Yes**
Account Name	Specifies the name of the account that the phone displays on the phone screen.
SIP Server	Specifies the IP address and port of the SIP server on which you created the account.
Secondary SIP Server	Specifies the IP address and port of the SIP server that the deskphone uses when the primary SIP server fails.

Table continues…

Name	Description
Voice Mail UserID	Specifies the number of the voice mail server that the deskphone dials to when the user presses the **Message** button.

Related links

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Configuring the accounts network settings

Procedure

1. Click **Accounts** > **Account 1** > **Network Settings**.

   The system displays the Network Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Accounts network settings field descriptions

Name	Description
DNS Mode	Specifies the method that the system uses to look up IP addresses for host names. You can select from the following options: • A Record • SRV • NAPTR/SVR • Use Configured IP If you select `Use Configured IP`, you must also provide values for **Primary IP**, **Backup IP 1**, and **Backup IP 2**.
Primary IP	Specifies the primary IP address to which the phone sends DNS queries.
Backup IP 1	Specifies the first backup IP address to which the phone sends DNS queries if the primary IP address fails.

Table continues…

Name	Description
Backup IP 2	Specifies the second backup IP address to which the phone sends DNS queries if the first backup IP address fails.
Proxy-Require	Specifies the SIP extension to notify the SIP server that the phone is behind a NAT or a firewall.

Related links

Administering through the Web GUI on page 19

Configuring the accounts SIP settings

Configuring the basic SIP settings

Procedure

1. Click **Accounts** > **Account 1** > **SIP Settings** > **Basic Settings**.

   The system displays the Basic Settings page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Basic SIP settings field description

Name	Description
SIP Registration	Specifies whether the deskphone sends SIP register messages to the proxy server. Select from the following options: • **No**: The deskphone does not send SIP register messages • **Yes**: The deskphone sends SIP register messages
Unregister on Reboot	Specifies whether the deskphone clears the SIP registration information when the deskphone reboots. Select from the following options: • **No**: To keep the information

Table continues…

Name	Description
	• **Yes**: To clear the information If you enabled the auto login feature, the feature takes precedence over any value set in this field.
Register Expiration	Specifies the frequency in minutes in which the deskphone refreshes registration with the specified registrar. The maximum value that you can assign is 64800. The default value is 60.
Reregister before Expiration	Specifies the frequency in seconds in which the deskphone sends reregistration request before the deskphone refreshes its registration. The default value is 0.
SIP Registration Failure Retry Wait Time	Specifies the interval in seconds in which the deskphone retries registration if the registration process fails. Assign any value in the range of 1 to 3600. The default values is 20.
SIP T1 Timeout	Specifies the SIP T1 timeout, which is an estimate of the round-trip time (RTT) of transactions between the deskphone and the SIP server. The default is 0.5 seconds.
SIP T2 Timeout	Specifies the SIP T2 timeout, which is the maximum retransmit interval for non-INVITE requests and INVITE responses. The default setting is 4 seconds.
SIP Transport	Specifies the network protocol that the deskphone uses for the SIP transport. Select from the following options: • **TCP** • **TLS/TCP**
SIP URI Scheme When Using TLS	Specifies the SIP URI scheme that the deskphone uses when you select TLS/TCP as the SIP transport. Select from the following options: • **sip** • **sips** **Note:** In case of TLS registration, you must select **sip** so that the deskphone can register and make or receive calls behind Avaya SBCE (Session Border Controller for Enterprise) over TLS.
Use Actual Ephemeral Port in Contact with TCP/TLS	Specifies whether the deskphone uses the actual ephemeral port in contact when you select TLS/TCP

Table continues…

Name	Description
	as the SIP transport. Select from the following options: • **No** • **Yes** **Note:** Avaya E129 SIP Deskphone does not support this parameter. Ensure that you select **No**.
Remove OBP from Route	Specifies whether the deskphone notifies the SIP server of being behind a NAT or a firewall. Select from the following options: • **No** • **Yes**
Support SIP Instance ID	Specifies whether the deskphone supports the SIP Instance ID. Select from the following options: • **No** • **Yes**
SUBSCRIBE for MWI	Specifies whether the deskphone sends SUBSCRIBE for Message Waiting Indication periodically. Select from the following options: • **No** • **Yes**
SUBSCRIBE for Registration	Specifies whether the deskphone sends SUBSCRIBE for registration periodically. Select from the following options: • **No** • **Yes**
Enable 100rel	Specifies whether the deskphone invokes a reliable provisional response by appending the 100rel tag to the value of the required header of the initial signaling messages. Select from the following options: • **No** • **Yes**
Caller ID Display	Specifies whether the deskphone displays the caller ID. You can select from the following options: • **Auto**: The deskphone looks for the caller ID in the order of the P-Asserted Identity Header field, the

Table continues…

Name	Description
	Remote-Party-ID Header field, and the From Header field in the incoming SIP INVITE . • **Disabled**: The deskphone displays unavailable instead of the caller ID. • **From Header**: The deskphone displays the caller ID based on the From Header field in the incoming SIP INVITE.
Use Privacy Header	Specifies whether the SIP INVITE message includes the Privacy Header field. Select from the following options: • **Default**: To not include the Privacy Header field when you activate the Huawei IMS feature. • **No**: To not include the Privacy Header field. • **Yes**: To include the Privacy Header field.
Use P-Preferred-Identity Header	Specifies whether the SIP INVITE message includes the P-Preferred-Identity Header field. Select from the following options: • **Default**: Includes the P-Preferred-Identity Header field when you activate the Huawei IMS feature. • **No**: Includes the P-Preferred-Identity Header field. • **Yes**: Includes the P-Preferred-Identity Header field.

Related links

Administering through the Web GUI on page 19

Configuring the advanced SIP settings

About this task

Note:

For now, Avaya E129 SIP Deskphone does not support advanced SIP settings.

Procedure

1. Click **Accounts** > **Account 1** > **SIP Settings** > **Advanced Features**.

   The system displays the Advanced Features page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Configuring the SIP session timer settings

Procedure

1. Click **Accounts** > **Account 1** > **SIP Settings** > **Session Timer**.

   The system displays the Session Timer page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

SIP session timer settings field descriptions

Name	Description
Session Expiration	Specifies the session expiration time in seconds provided that no successful session refresh transaction occurs beforehand.
Min-SEe	Specifies the minimum session expiration time in seconds.
Caller Request Timer	Specifies whether the phone uses a session timer while making outbound calls when the remote party supports session timers.
Callee Request Timer	Specifies whether the phone uses a session timer while receiving inbound calls when the remote party supports session timers.
Force Timer	Specifies whether the phone uses a session timer even if the remote party does not support this feature.
UAC Specify Refresher	Specifies the refresher to select as a caller. You can select from the following options: • UAC: To use the phone as the refresher. • UAS: To use the callee or proxy server as the refresher. • Omit . As a Caller, select UAC to use the phone as the refresher; or select UAS to use the Callee or proxy server as the refresher.

Table continues…

Name	Description
UAS Specify Refresher	Specifies the refresher to select as a callee. You can select from the following options: • UAC: To use caller or proxy server as the refresher. • UAS: To use the phone as the refresher.
Force INVITE	Specifies whether the phone uses the INVITE method to refresh the session timers.

Related links

[Administering through the Web GUI](#) on page 19

Configuring the SIP security settings

Procedure

1. Click **Accounts** > **Account 1** > **SIP Settings** > **Security Settings**.

   The system displays the Security Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI](#) on page 19

SIP security settings field descriptions

Name	Description
Check Domain Certificates	Specifies whether the phone checks the domain certificates when the SIP transport is TLS or TCP.
Validate Incoming Messages	Specifies whether the phone validates the incoming messages.
Check SIP User ID for Incoming INVITE	Specifies whether the phone checks the SIP user ID in the request URI of the incoming INVITE. If the SIP user ID does not match the phone SIP user ID, the phone rejects the call.
Accept Incoming SIP from Proxy Only	Specifies whether the phone checks the SIP address of the request URL in the incoming SIP message. If the SIP address of the request URL doe snot match with the SIP server address of the account, the phone rejects the call.

Table continues…

Name	Description
Authenticate Incoming INVITE	Specifies whether the phone challenges the incoming INVITE for authentication with SIP 401 Unauthorized response.

Related links

Administering through the Web GUI on page 19

Configuring the accounts audio settings

Procedure

1. Click **Accounts** > **Account 1** > **Audio Settings**.

   The system displays the Audio Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Accounts audio settings field descriptions

Name	Description
Send DTMF	Specifies the mechanism to transmit DTMF digits. You can select from the following options: • in-audio • via RTP (RFC2833) • via SIP INFO
DTMF Payload Type	Specifies the payload type for DTMF as specified in RFC2833.
Preferred Vocoder - choice 1	Specifies the first preferred vocoders that the phone includes in the same preference order in the SDP message.
Preferred Vocoder - choice 2	Specifies the second preferred vocoders that the phone includes in the same preference order in the SDP message.

Table continues…

Name	Description
Preferred Vocoder - choice 3	Specifies the third preferred vocoders that the phone includes in the same preference order in the SDP message.
Preferred Vocoder - choice 4	Specifies the fourth preferred vocoders that the phone includes in the same preference order in the SDP message.
Preferred Vocoder - choice 5	Specifies the fifth preferred vocoders that the phone includes in the same preference order in the SDP message.
Preferred Vocoder - choice 6	Specifies the sixth preferred vocoders that the phone includes in the same preference order in the SDP message.
Preferred Vocoder - choice 7	Specifies the seventh preferred vocoders that the phone includes in the same preference order in the SDP message.
Use First Matching Vocoder in 200OK SDP	Specifies whether the phone uses the first matching vocoder as the codec that the phone received in the 200OK SDP.
SRTP Mode	Specifies whether the SRTP mode is to be enabled.
Symmetric RTP	Specifies whether the phone supports the symmetric RTP.
Silence Suppression	Specifies whether the phone sends a small quantity of VAD packets, instead of audio packets, when the phone detects silence. Silence suppression is applicable for the codec G.723 and G.729 only.
Voice Frames per TX	Specifies the number of voice frames that the phone tramsits per packet. the number of voice frames transmitted per packet. The frame number is up to 10, 20, 32, and 64 for G711, G726, G723, and other codecs respectively.
G723 Rate	Specifies the encoding rate for G723 codec.
G.726-32 Packing Mode	Specifies the packing mode for the G726-32 codec.
iLBC Frame Size	Specifies the iLBC packet frame size.
iLBC Payload Type	Specifies the iLBC payload type. The valid range is from 96 to 127.
Jitter Buffer Type	Specifies the jitter buffer type. You can select from the following values based on network conditions: • Fixed • Adaptive
Jitter Buffer Length	Specifies jitter buffer length. You can select a low, medium, or high value based on network conditions.

Related links

Administering through the Web GUI on page 19

Configuring the accounts call settings

Procedure

1. Click **Accounts** > **Account 1** > **Call Settings**.

   The system displays the Call Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Accounts call settings field descriptions

Name	Description
Early Dial	Specifies whether the SIP proxy supports the 484 response.
Dial Plan Prefix	Specifies the prefix that the phone adds to the dialed number.
Dial Plan	Specifies the dial plan for the phone. The dial plan establishes the expected number and pattern of digits for a telephone number. You can use the following rules to establish the dial plan: • Accepted digits and characters: 0 to 9, *, #, A to D, a to d. • x: Represents any digit from 0 to 9. • xx+: Represents at least a 2 digit number. • ^: Excludes the following digits or characters. • [x-y]- Includes any digit from the range x to y. • <x=yz>: Replaces digit x with digits y and z when dialing. • |: Represents the Or operation. • ,: Use comma for adding a secondary dial tone in the dial plan. Comma takes the higher precedence over Or in the dial plan. Use of caret ^ is not

Table continues…

Name	Description
	supported after comma. You cannot replace digits after a comma. For example, {^1900x+ | <=1617>xxxxxxx} that specifies to block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers.
Delayed Call Forward Wait Time	Specifies the timeout in seconds before the phone forwards the call on no answer. You can specify any value from 1 to 120.
Enable Call Features	Specifies whether the DND, call forward, and other features can be set through the local feature code on the phone or through the ITSP feature code. **Note:** Avaya E129 SIP Deskphone does not support this parameter.
Call Log	Specifies the call log settings for the phone.
Account Ring Tone	Specifies the ring tone for the account.
Match Incoming Caller ID	Specifies the ring tone that the phone plays when an incoming caller ID matches the specified rule. You can specify the matching rules through: • A specific caller ID number, such as 8321123. • A defined pattern with certain length using x and + to specify a caller Id, where x is any digit from 0 to 9. For example, [345]xx specify a 3–digit number with the leading digit as 3, 4, or 5. • An Alert Info text, such as priority. The phone plays the selected ring tone if the phone receives a SIP INVITE with the Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority.
Ring Timeout	Specifies the timeout in seconds for rings on no answer. You can specify any value in the range of 30 to 3600.
Send Anonymous	Specifies whether the From header in the outgoing INVITE messages is set to anonymous.
Anonymous Call Rejection	Specifies whether the phone rejects the anonymous calls.
Auto Answer	Specifies whether the phone automatically turns on the speaker phone to answer incoming calls after a short reminding beep.
Refer-To Use Target Contact	Specifies whether the Refer-To header uses the Contact header information of the transferred target for attended transfer.

Table continues…

Name	Description
Transfer on Conference Hangup	Specifies whether the phone transfers the call to another party if the conference initiator hangs up.
No Key Entry Timeout	Specifies the timeoutout in seconds when the phone dials the digits if no key is pressed after the timeout.
Use # as Dial Key	Specifies whether the # key is used to immediately dial out the input digits.

Related links

Administering through the Web GUI on page 19

Managing the deskphone settings

Configuring the general settings

Procedure

1. Click **Settings** > **General Settings**.

   The system displays the General Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

General settings field descriptions

Name	Description
Local RTP Port	Specifies the local RTP port that the deskphone uses to listen and transmit. The value must be an even number and in the range of 1024 to 65400.
Use Random Port	Specifies whether the deskphone forces a random generation of both the local SIP and RTP ports.
Enable auto login	Specifies whether the auto login feature is turned on or off.

Related links

Administering through the Web GUI on page 19

Configuring the call features settings

Procedure

1. Click **Settings** > **Call Features**.

   The system displays the Call Features page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Call features settings field descriptions

Name	Description
Off-hook Auto Dial	Specifies the number that the deskphone dials when the deskphone is off-hook.
Off-hook Timeout	Specifies the time in seconds after which the deskphone goes on-hook after being off-hook. You can specify any value in the range of 10 to 60.
Disable Call Waiting	Specifies whether to disable the call waiting feature.
Disable Call Waiting Tone	Specifies whether to disable the call waiting tone when the Call Waiting feature is active.
Use Quick IP Call Mode	Specifies whether the user can dial an IP address under the same LAN or VPN segment by entering the last octet in the IP address. **Note:** Avaya E129 SIP Deskphone does not support this parameter.
Disable in-call DTMF Display	Specifies whether the deskphone displays the DTMF digits that the user enters during a call.
Enable Sending DTMF via Speed Dial	Specifies whether to enable Speed Dial Key to send DTMF during a call.
In-call Dial Number on Pressing Transfer Key	Specifies the number that the deskphone dials as DTMF when the user presses the Transfer button.
Auto-Attended Transfer	Specifies whether the deskphone uses the attended transfer by default.
Do Not Escape '#' as %23 in SIP URI	Specifies whether the deskphone replaces # by %23 for some special situations.

Table continues…

Name	Description
Click-To-Dial Feature	Specifies whether to enable the Click-To-Dial feature. **Note:** Avaya E129 SIP Deskphone does not support this parameter.
Blink message LED on ringing	Specifies whether the deskphone blinks the Message LED for an incoming call.
Call History Flash Writing	Specifies the interval in seconds for the deskphone to save the call history to the flash memory. The default value is 300 seconds and 0 implies that the option is disabled.
Write Timeout	Specifies the interval in seconds for the deskphone to save the call history to the flash memory. You can specify any value in the rage of 30 to 3600.
Max Unsaved Log	Specifies the number of unsaved logs before the deskphone writes them to the flash memory. You can specify any value in the rage of 0 to 500.

Related links

Administering through the Web GUI on page 19

Configuring the ring tone settings

Procedure

1. Click **Settings** > **Ring Tone**.

   The system displays the Ring Tone page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Ring tone settings field descriptions

Name	Description
Call Progress Tones	Displays the syntax of specifying the frequency and cadence for ring tones that the deskphone plays at various instances.

Table continues…

Name	Description
	Use the following syntax to specify frequencies and cadence for a ring tone. f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; where • f1 and f2 are frequencies in Hz. • c is cadence on and off in ms. The system supports up to three cadences. For example, f1=440,f2=480,c=200/400;
System Ring Tone	Specifies frequency and cadence for system ring tone.
Dial Tone	Specifies frequency and cadence for dial tone.
Message Waiting	Specifies frequency and cadence for message waiting.
Ring Back Tone	Specifies frequency and cadence for ring back tone.
Call Waiting Tone Gain	Specifies frequency and cadence for call waiting tone gain.
Busy Tone	Specifies frequency and cadence for busy tone.
Reorder Tone	Specifies frequency and cadence for reorder tone.

Related links

Administering through the Web GUI on page 19

Configuring the audio control settings

Procedure

1. Click **Settings** > **Audio Control**.

   The system displays the Audio Control page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Audio control settings field descriptions

Name	Description
Headset	

Table continues…

Name	Description
HEADSET Key Mode	Specifies the mode for the headset button when a headset is connected to the deskphone. You can select from the following options: • Default mode. In the Default mode: - Pressing **Headset** when the deskphone is idle puts the deskphone off-hook, and the deskphone starts using the headset for calls. - Pressing **Headset** when an incoming call comes answers the call through the headset. - Pressing **Headset** during an active call through the headset disconnects the call. - Pressing **Headset** when the speaker or headset is used for dialing or talking switches the deskphone to the headset. Pressing **Headset** again disconnects the call. Pressing **Speaker** switches back to the previous mode. • Toggle Headset/Speaker mode. In the Toggle Headset/Speaker mode: - Pressing **Headset** when the deskphone is idle switches the deskphone to the Headset mode. In the Headset mode, pressing **Speaker** or the line key puts the deskphone off-hook, and the deskphone starts using the headset for calls. - Pressing **Headset** during an active call changes between the headset and speaker.
Headset Type	Specifies the headset type. You can select from the following types: • Normal: An RJ9 headset • Plantronics EHS
Always Ring Speaker	Specifies whether to enable the speaker to ring when the deskphone uses the headset in the Toggle Headset/Speaker mode.
Headset TX Gain (dB)	Specifies the transmission gain of the headset.
Headset RX Gain (dB)	Specifies the receiving gain of the headset.
Handset	
Handset TX Gain (dB)	Specifies the transmission gain of the handset.

Related links

Administering through the Web GUI on page 19

Configuring the LCD display settings

Procedure

1. Click **Settings** > **LCD Display**.

   The system displays the LCD Display page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

LCD display settings field description

Name	Description
LCD Contrast	Specifies the LCD contrast level. You can specify any value in the range of 0 to 20.

Related links

Administering through the Web GUI on page 19

Configuring the date and time settings

Procedure

1. Click **Settings** > **Date and Time**.

   The system displays the Date and Time page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Date and time settings field descriptions

Name	Description
NTP Server	Specifies the URL or IP address of the NTP server. The phone can obtain the date and time from the server.
Allow DHCP Option 42 to override NTP server	Specifies whether the DHCP Option 42 overrides the NTP server.
Time Zone	Specifies the time zone based on which the phone displays the date and time.
Allow DHCP Option 2 to Override Time Zone Setting	Specifies whether DHCP Option 2 overrides the time zone settings.
Self-Defined Time Zone	Specifies the user-defined time zone. You can use the following syntax to define the time zone: std offset dst [offset], start [/time], end [/time] The default is set to MTZ+6MDT+5,M4.1.0,M11.1.0 , where • MTZ+6MDT+5 indicates a time zone with six hours offset with one hour ahead of the U.S central time. Plus (+) indiactes the local time zone is west of the Prime Meridian. Minus (-) indicates the local time zone is east of the Prime Meridian. • M4.1.0,M11.1.0 in which the first number indicates the month, where 1 corresponds to January, 2 corresponds to February, and so on. The second number indicates the nth iteration of the weekday, where 1 corresponds to Sunday, 2 corresponds to Monday, and so on. The third number indicates weekday, where 0 corresponds to Sunday, 1 corresponds to Monday, and so on.
Date Display Format	Specifies the date display format on the LCD.
Time Display Format	Specifies the time display in 12-hour or 24-hour format on the LCD.

Related links

Administering through the Web GUI on page 19

Configuring the XML applications settings

Procedure

1. Click **Settings** > **XML Applications**.

The system displays the XML Applications page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI](#) on page 19

XML applications field descriptions

Name	Description
Idle Screen XML Download	Specifies whether to download the idle screen XML through HTTP, HTTPS, or TFTP.
Download Screen XML at Boot-up	Specifies whether the deskphone downloads the idle screen XML at the time of starting up.
Use Custom Filename	Optional. Specifies the name of the idle screen XML file that the deskphone downloads. Overrides the default file.
Idle Screen XML Server Path	Specifies the server path from where the phone downloads the idle screen XML file. You can specify an IP address or a URL with up to 256 characters.

Related links

[Administering through the Web GUI](#) on page 19

Managing the network settings

Configuring the network basic settings

Procedure

1. Click **Network** > **Basic Settings**.

   The system displays the Basic Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI](#) on page 19

Network basic settings field descriptions

Name	Description
Internet Protocol	Specifies Internet Protocol that the deskphone uses. Select from the following options: • **Prefer IPv4** • **Prefer IPv6** **Note:** Avaya E129 SIP Deskphone does not support IPv6.
IPv4 Address	Specifies the method for the deskphone to get the IPv4 address if you selected **Prefer IPv4** in the **Internet Protocol** field. Select from the following options: • **DHCP** • **PPPoE** • **Statically configured as** Depending on the method that you select, you must specify the respective fields. **Note:** Avaya E129 SIP Deskphone does not support PPPoE.

DHCP field descriptions

Specify the following fields if you selected **DHCP** in the **IPv4 Address** field.

Name	Description
Host name (Option 12)	Specifies the name of the client. This field is optional, but some Internet Service Providers might require it.
Vendor Class ID (Option 60)	Specifies vendor class ID that client and servers exchange.

Statically configured as field descriptions

Specify the following fields if you selected **Statically configured as** in the **IPv4 Address** field.

Name	Description
IPv4 Address	Specifies the IP address.
Subnet Mask	Specifies the subnet mask.
Gateway	Specifies the default gateway.

Table continues…

Name	Description
DNS Server 1	Specifies the first DNS server.
DNS Server 2	Specifies the second DNS server.
Preferred DNS Server	Specifies the preferred DNS server.

Related links

Administering through the Web GUI on page 19

Configuring the network advanced settings

Procedure

1. Click **Network** > **Advanced Settings**.

   The system displays the Advanced Settings page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Network advanced settings field descriptions

Name	Description
802.1X Mode	Specifies whether to enable the 802.1X mode. Supports only EAP-MD5.
802.1X Identity	Specifies the 802.1X mode.
MD5 Password	Specifies the MD5 password.
HTTP Proxy	Specifies the HTTP proxy URL where the phone sends the packets. The proxy server is an intermediary to route the packets to the destination.
HTTPS Proxy	Specifies the HTTPS proxy URL where the phone sends the packets. The proxy server is an intermediary to route the packets to the destination.
Layer 3 QoS	Specifies the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv, or MPLS.
Layer 2 QoS 802.1Q/VLAN Tag	Specifies the VLAN tag of the Layer 2 QoS packets.
Layer 2 QoS 802.1p Priority Value	Specifies the priority value of the Layer 2 QoS packets. You can specify any value in the range of 0 to 7.

Table continues…

Name	Description
PC port mode	Specifies the PC port mode. If you select `Mirrored`, the traffic in the LAN port goes through the personal computer port and the system can capture packets if you connect the computer to the PC port.
Enable LLDP	Specifies whether to enable the LLDP service. LLDP-MED is supported only for VLAN priority and Tag value. If Avaya-proprietary LLDP is not supported, use DHCP to receive file server or SIP server address parameters.

Related links

Administering through the Web GUI on page 19

Maintaining the deskphone

Configuring the Web access passwords

Procedure

1. Click **Maintenance** > **Web Access**.

   The system displays the Web Access page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

Web access passwords field descriptions

Name	Description
User password	
New Password	Specifies the user password. The password is case sensitive.
Confirm Password	Specifies the user password for confirmation.
Admin password	
New Password	Specifies the administrator password. The password is case sensitive.

Table continues…

Name	Description
	The default password is admin. You must change the default password.
Confirm Password	Specifies the administrator password for confirmation.

Related links

Administering through the Web GUI on page 19

Configuring the upgrade and provisioning settings

About this task

Use the following procedure to upgrade the firmware of the deskphone. During the upgrade, if the deskphone receives a call, the deskphone starts ringing. If the user does not answer the call, the deskphone continues to ring till the upgrade completes. Therefore, you must plan the upgrade during the off hours.

Procedure

1. Click **Maintenance** > **Upgrade and Provisioning**.

   The system displays the Upgrade and Provisioning page.
2. Enter the required details for each field.
3. Click **Save**.
4. Click **Reboot** at the upper-right corner of the page for the changes to take effect.

Related links

Administering through the Web GUI on page 19

Upgrading and provisioning settings field descriptions

Name	Description
Firmware Upgrade and Provisioning	Specifies whether the deskphone checks and installs a new firmware every time the deskphone boots up: The options are: • **Always Check for New Firmware**: The deskphone always checks for a new firmware. • **Check New Firmware Only When F/W pre/suffix Changes**: The deskphone always checks for a change in the prefix or suffix of the firmware file.

Table continues…

Name	Description
	• **Always Skip the Firmware Check**: The deskphone does not check for a new firmware.
XML Config File Password	Specifies the password that the deskphone uses to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	Specifies the user name to login to the HTTP and HTTPS server.
HTTP/HTTPS Password	Specifies the password to login to the HTTP and HTTPS server.
Upgrade via	Specifies the firmware upgrade method. The options are: • **TFTP** • **HTTP** • **HTTPS**
Firmware Server Path	Specifies the server path for the firmware.
Config Server Path	Specifies the sever path for the configuration file.
Firmware File Prefix	Specifies the prefix for the firmware file. The deskphone downloads the firmware file only if the prefix matches the one that you specified.
Firmware File Postfix	Specifies the postfix for the firmware file. The deskphone downloads the configuration file only if the postfix matches the one that you specified.
Config File Prefix	Specifies the prefix for the configuration file. The deskphone downloads the configuration file only if the prefix matches the one that you specified.
Config File Postfix	Specifies the postfix for configuration file. The deskphone downloads the configuration file only if the postfix matches the one that you specified.
Allow DHCP Option 43 and Option 66 to Override Server	Specifies whether the DHCP option 66 overrides the configuration server. If you enable the DHCP option 66, the system can redirect the HTTP/TFTP server. The options are: • **No** • **Yes**
Allow DHCP Option 120 to Override SIP Server	Specifies whether the DHCP Option 120 from local server overrides the SIP server. The options are: • **No** • **Yes**

Table continues…

Name	Description
	Note: You must use the configuration file to enter the SIP server list as the DHCP Option 120 has certain limitations.
Automatic Upgrade	Specifies whether the deskphone checks and installs a new firmware. The options are: • **No**: The deskphone does not check for a new firmware. • **Yes, check for upgrade every x minute(s)**: The deskphone checks for a new firmware after every x minutes. • **Yes, check for upgrade every day**: The deskphone checks for a new firmware every day. • **Yes, check for upgrade every week**: The deskphone checks for a new firmware every week. Enabling the automatic upgrade is a preferred option as the deskphone gets a firmware upgrade without you rebooting the deskphone for an upgrade.
Hour of the Day(0-23)	Specifies the hour of the day when the deskphone checks the HTTP/TFTP server for a firmware upgrade or configuration file changes.
Day of the Week (0-6)	Specifies the day of the week when the deskphone checks the HTTP/TFTP server for a firmware upgrade or configuration file changes.
Authenticate Conf File	Specifies whether the deskphone authenticates the configuration file before accepting the file. The options are: • **No** • **Yes**

Related links

Administering through the Web GUI on page 19

Configuring the syslog settings

About this task

You must reboot the deskphone to apply the syslog settings.

Procedure

1. Click **Maintenance** > **Syslog**.

The system displays the Syslog page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI](#) on page 19

Syslog settings field descriptions

Name	Description
Syslog Server	Specifies the URL or IP address of the syslog server.
Syslog Level	Specifies the level of syslog logging.
Send SIP Log	Specifies whether the system includes the SIP log in the syslog messages.
Auto Recover from Abnormal	Specifies whether the phone autorecovers when it does not run normally.

Related links

[Administering through the Web GUI](#) on page 19

Configuring the language settings

Procedure

1. Click **Maintenance** > **Language**.

   The system displays the Language page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI](#) on page 19

Language settings field descriptions

Name	Description
Display Language	Specifies the display language on the phone.
Language File Postfix	Specifies the language file postfix for the language that the phone downwloads.

Related links

[Administering through the Web GUI] on page 19

Configuring the TR-069 settings

About this task

Note:

Avaya E129 SIP Deskphone does not support TR-069 settings.

Procedure

1. Click **Maintenance** > **TR-069**.

   The system displays the TR–069 page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI] on page 19

Configuring the security settings

Procedure

1. Click **Maintenance** > **Security**.

   The system displays the Security page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

[Administering through the Web GUI] on page 19

Security settings field descriptions

Name	Description
Configuration via Keypad Menu	Specifies the access control for keypad Menu settings. You can select from the following options: • **Unrestricted**: Displays all menu options on the deskphone.

Table continues…

Name	Description
	• **Basic settings only**: Hides the Config menu option on the deskphone. • **Constraint Mode**: Hides the Config, Factory Functions, and Network Config menu options on the deskphone.
Enable STAR Key Keypad Locking	Specifies whether holding the star (*) key for 4 seconds locks the keypad.
Password to Lock/Unlock	Specifies the password to lock and unlock the keypad.
SIP TLS Identity certificate	Specifies the identity certificate file for the phone to authenticate over the SIP server through TLS. Supports the PEM file format only.
SIP TLS Identity certificate private key	Specifies the private key for the SIP TLS identity certificate in PEM format.
SIP TLS Identity certificate private key passphrase	Specifies the passphrase of the private key for the SIP TLS identity certificate in text format.
Download Device Configuration	Downloads the phone configuration file in .txt format.
Web Access Mode	Specifies the protocol for the Web interface.
Disable SSH	Specifies whether to disable SSH access.

Related links

Administering through the Web GUI on page 19

Managing the phone book

Adding a contact

Procedure

1. Click **Phonebook** > **Contacts**.

   The system displays the Contacts page.
2. Click **Add Contact**.

   The system displays the Add Contact dialog box.
3. Enter the required details for each field.
4. Click **Save**.

Related links

Administering through the Web GUI on page 19

Updating contact details

Procedure

1. Click **Phonebook** > **Contacts**.

   The system displays the Contacts page.
2. Click **Edit Contact** for the respective contact that you want to update.

   The system displays the Edit Contact dialog box.
3. Change the details of the required fields.
4. Click **Save**.

Related links

Administering through the Web GUI on page 19

Deleting a contact

Procedure

1. Click **Phonebook** > **Contacts**.

   The system displays the Contacts page.
2. Click **Edit Contact** for the respective contact that you want to delete.

   The system displays the Edit Contact dialog box.
3. Click **Delete**.
4. Click **OK** to confirm the deletion.

Related links

Administering through the Web GUI on page 19

Contact field descriptions

Name	Description
First Name	Specifies the first name of the contact. You can enter up to 24 characters.
Last Name	Specifies the last name of the contact. You can enter up to 24 characters.
Phone Number	Specifies the phone number of the contact.
Accounts	Specifies the account of the contact.
Groups	Specifies the group to which the contact belongs.

Related links

Administering through the Web GUI on page 19

XML phone book

The Avaya E129 SIP Deskphone supports an XML-based phone book that you can store and maintain on a Web server. You can also download the XML phone book on the deskphone and upload it on the Web server. You can use the Web GUI to download and upload the phone book. You can also set up a centralized directory server with which the phones can synchronize periodically. The Avaya E129 SIP Deskphone supports a maximum of 2000 contacts.

Before you use the XML phone book, you must upgrade the deskphone to the latest firmware. Go to the Avaya Support website for the latest firmware.

Related links

Administering through the Web GUI on page 19
XML phone book structure on page 50
Configuring the phone book settings on page 52
Phone book settings field descriptions on page 52

XML phone book structure

```
<?xml version="1.0" encoding="UTF-8"??\>
    <AddressBook>
        <Contact>
            <LastName>The last name of the contact</LastName>
            <FirstName>The first name of the contact</FirstName>
            <Phone>
                <phonenumber>The phone number of the contact</phonenumber>
                <accountindex>The account index number</accountindex>
            </Phone>
            <Groups>
                <groupid>The group ID of the contact</groupid>
            <Groups>
        </Contact>

        <Contact>
            <!-- Enter the details of another contact. Follow the same format as the
previous one. -->
        </Contact>
    </AddressBook>
```

<AddressBook> element

Object	Position	Mandatory	Type	Description
AddressBook	Root element	Yes	-	Root element of the XML document.
Contact	Child element	Yes	-	Each contact element corresponds to a phone book entry.

<Contact> element

Object	Position	Mandatory	Type	Description
Contact	Element	Yes		
LastName	Child element	Either FirstName or the LastName element is present	String	The last name of the contact.
FirstName	Child element	Either FirstName or the LastName element is present	String	The first name of the contact.
Phone	Child element	Yes	-	The phone number and the account index.
Groups	Child element	No	-	The group ID of the contact.

<Phone> element

Object	Position	Mandatory	Type	Description
Phone	Element	Yes	-	Root element of the XML document.
phonenumber	Child element	Yes	int	The phone number of the contact.
accountindex	Child element	Yes	int	The account index.

<Groups> element

Object	Position	Mandatory	Type	Description
Groups	Element	No	-	The group ID of the contact.
groupid	Child element	Yes	int	Valid values are from 0 to 9, where: • 0: Family • 1: Friends • 2: Work The values 3 to 9 are the user-entered values.

XML phone book example

```
<?xml version="1.0" encoding="UTF-8"??\>
    <AddressBook>
        <Contact>
            <LastName>Doe</LastName>
            <FirstName>John</FirstName>
            <Phone>
                <phonenumber>8000</phonenumber>
                <accountindex>1</accountindex>
            </Phone>
            <Groups>
                <groupid>0</groupid>
```

```
        <Groups>
    </Contact>

    <Contact>
        <LastName>Smith</LastName>
        <FirstName>Alan</FirstName>
        <Phone>
            <phonenumber>8001</phonenumber>
            <accountindex>1</accountindex>
        </Phone>
        <Groups>
            <groupid>1</groupid>
        <Groups>
    </Contact>

    <Contact>
        <LastName>Lee</LastName>
        <FirstName>Lily</FirstName>
        <Phone>
            <phonenumber>6000</phonenumber>
            <accountindex>2</accountindex>
        </Phone>
        <Groups>
            <groupid>2</groupid>
        <Groups>
    </Contact>

</AddressBook>
```

Related links

XML phone book on page 50

Configuring the phone book settings

Procedure

1. Click **Phonebook** > **Phonebook Management**.

   The system displays the Phonebook Management page.

2. Enter the required details for each field.
3. Click **Save**.

Related links

XML phone book on page 50

Phone book settings field descriptions

Name	Description
Enable Phonebook XML Download	Specifies whether to enable the XML phone book download. Select one of the following options: • **Disabled** • **Enabled, use TFTP** • **Enabled, use HTTP**

Table continues…

Name	Description
	• **Enabled, use HTTPS**
Phonebook XML Server Path	Specifies the server path, IP address or URL, to download the XML phone book XML. You can enter up to 256 characters. Use one of the following syntax to provide the server path: • IP_address[:port]/dir • Hostname[:port]/dir For example, 192.168.40.10/XMLphone bookdir service.mycompany.com/XML/phone book If you do not specify the port, the system uses port 80 as a default port for HTTP, port 443 as a default port for HTTPS, and port 69 as a default port for TFTP. Do not specify the phone book file name in the server path, but only the directory path as Avaya E129 SIP Deskphone looks for the `(null)_phonebook.xml` file to download the phone book. If you enter the server path in the browser, the browser displays the XML phone book. You can save the XML phone book on your personal computer.
Phonebook Download Interval	Specifies the phone book download interval in minutes. 0 disables the automatic download. You can specify any value in the range of 5 to 720.
Remove Manually-edited Entries on Download	Specifies whether the phone deletes the manual entries when the phone downloads the XML phone book.
Download XML Phonebook	Downloads the XML phone book from the deskphone to the personal computer.
Upload XML Phonebook	Uploads the XML phone book from the personal computer to the deskphone. If you modify a contact in the XML phone book and upload the XML phone book on the deskphone, the deskphone creates a new entry for the modified contact and keeps the previous one as it is.

Table continues…

Name	Description
Phonebook Key Function	Specifies the **Phonebook** key behavior. Select one of the following options: • **Default** • **LDAP Search** • **Local Phonebook** • **Local Group**

Related links

XML phone book on page 50

Configuring the LDAP settings

Procedure

1. Click **Phonebook** > **LDAP**.

   The system displays theLDAP page.
2. Enter the required details for each field.
3. Click **Save**.

Related links

Administering through the Web GUI on page 19

LDAP settings field descriptions

Name	Description
Server Address	Specifies the IP address or DNS name of the LDAP server.
Port	Specifies the port of the LDAP server.
Base	Specifies the LDAP search base. The search base is location in the directory from where the search begins. For example dc=grandstream, dc=com.
User Name	Specifies the bind user name for querying the LDAP server. If the LDAP server allows anonymous binds, you can leave this field blank.
Password	Specifies the bind password for querying the LDAP server. If the LDAP server allows anonymous binds, you can leave this field blank.

Table continues…

Name	Description
LDAP Number Filter	Specifies the filter that the phone uses for number lookups. For example, (|(telephoneNumber=%)(Mobile=%) returns all records which has the telephone number field or mobile field starting with the prefix that you enter.
LDAP Name Filter	Specifies the filter that the phone uses for name lookups. For example, (|(cn=%)(sn=%)) returns all records which has the cn field or sn field starting with the prefix that you enter.
LDAP Version	Specifies the protocol version for the phone to send the bind requests.
LDAP Name Attributes	Specifies the space-separated name attributes that the phone returns for an LDAP search. For example, cn sn description.
LDAP Number Attributes	Specifies the space-separated number attributes that the phone returns for an LDAP search. For example, telephoneNumber Mobile.
LDAP Display Name	Specifies space-separated attributes that the phone displays. You can display up to 3 attributes. For example, %cn %sn %telephoneNumber.
Max. Hits	Specifies the maximum number of results that an LDAP search returns. You can specify any value in the range of 1 to 3000.
Search Timeout	Specifies the interval in seconds for the server to process the request and for the client to wait for the sever result. You can specify any value in the range of 0 to 180.
Sort Results	Specifies whether the phone sorts LDAP search result.
LDAP Lookup	Specifies whether to enable number searching at the time of dialing and receiving calls.
Lookup Display Name	Specifies the space-separated display name for the attributes that an LDAP search looks up for incoming and outgoing calls. The values must be a subset of values that you enter for the **LDAP Name Attributes** field.

Related links

Administering through the Web GUI on page 19

Chapter 5: Administering through the configuration file

You can administer the deskphone centrally through the configuration file that Avaya provides with E129 deskphones. The configuration file resides on the configuration server and is a binary or an XML file that contains configuration parameters. Each parameter corresponds to a feature that you configure through the Web interface. To configure a feature, you can assign the required value to the parameter of that feature.

When the deskphone starts or restarts, the deskphone makes a TFTP, HTTP, or HTTPS request to the configuration server for configuration files. The deskphone requests for an XML file named cfgxxxxxxxxxxxx.xml followed by a file named cfgxxxxxxxxxxxx, where xxxxxxxxxxxx is the MAC address of the deskphone. For example, cfg000b820102ab.xml and cfg000b820102ab. If these files are unavailable, the deskphone downloads a generic configuration file, cfg.xml. The configuration file name must be in lowercase.

Related links

Account general settings parameters on page 57
Account network settings parameters on page 57
Basic SIP settings parameters on page 58
Advanced SIP settings parameters on page 60
SIP session timer settings parameters on page 61
SIP security settings parameters on page 63
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Call settings parameters on page 67
Basic network settings parameters on page 71
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Language parameters on page 77
Security settings parameters on page 78
XML phone book configuration parameters on page 79
LDAP settings parameter on page 80
General settings parameters on page 81
Call features settings parameters on page 82

Account general settings parameters

Parameter	Default value	Description
P271	1	Specifies whether to keep the account active. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P270		Specifies the account name.
P47		Specifies the URL or IP address of the SIP server. The supported syntax is `IP/Host:port`.
P2312		Specifies the URL or IP address of the secondary SIP server.
P3		Specifies the display name.
P33		Specifies the voice mail User ID.

Related links

Administering through the configuration file on page 56

Account network settings parameters

Parameter	Default value	Description
P103	0	Specifies the DNS mode. You can assign one of the following values to this parameter: • 0: A record • 1: SRV • 2: NAPTR/SRV • 3: Use configured IP

Table continues…

Parameter	Default value	Description
P2308		Specifies the primary IP. The maximum length of the parameter is 15 characters.
P2309		Specifies the first backup IP.
P2310		Specifies the second backup IP.
P197		Specifies a SIP extension to enbale firewall penetration. The maximum length of the parameter is 64 characters.

Related links

Administering through the configuration file on page 56

Basic SIP settings parameters

Parameter	Default value	Description
P63	0	Specifies the TEL URL. Assign one of the following values to this parameter: • 0: Disabled • 1: User=Phone • 2: Enabled
P31	1	Specifies the SIP registration. Assign one of the following values to this parameter: • 0: No • 1: Yes
P81	0	Specifies whether the phone unregisters after restarting. Assign one of the following values to this parameter: • 0: No • 1: Yes
P32	60	Specifies the time in minutes in which the registration expires. Assign any value in the range of 1 to 64800.
P2330	0	Specifies the time in seconds in which the phone reregisters before the registration expiration. Assign any value in the range of 0 to 64800.
P138	20	Specifies the SIP registration failure retry wait time in seconds. Assign any value in the range of 1 to 3600.
P209	50	Specifies the SIP T1 timeout. Assign one of the following values to this parameter: • 50 • 100

Table continues…

Parameter	Default value	Description
		• 200
P250	400	Specifies the SIP T2 interval. Assign one of the following values to this parameter: • 200 • 400 • 800
P130	0	Specifies the SIP transport protocol. Assign one of the following values to this parameter: • 1: TCP • 2: TLS/TCP
P2329	1	Specifies the SIP URI scheme when TLS is used. You can assign one of the following values to this parameter: • 0: sip • 1: sips
P2331	0	Specifies whether to use the actual ephemeral port in contact with TCP/TLS. Aassign one of the following values to this parameter: • 0: No • 1: Yes
P2305	0	Specifies whether to remove OBP from route. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P288	1	Specifies whether to support SIP instance ID. Assign one of the following values to this parameter: • 0: No • 1: Yes
P99	0	Specifies whether to subscribe for MWI. Assign one of the following values to this parameter: • 0: No • 1: Yes
P2319	1	Specifies whether to subscribe for registration. Assign one of the following values to this parameter: • 0: No • 1: Yes

Table continues…

Parameter	Default value	Description
P272	0	Specifies whether to enable 100rel. Assign one of the following values to this parameter: • 0: No • 1: Yes
P2324	0	Specifies whether to display caller ID. Assign one of the following values to this parameter: • 0: Auto • 1: Disabled • 2: From Header
P2338	0	Specifies whether to use privacy header. Assign one of the following values to this parameter: • 0: Default • 1: No • 2: Yes
P2339	0	Specifies whether to use P-Preferred-Identity header. Assign one of the following values to this parameter: • 0: Default • 1: No • 2: Yes

Related links

Administering through the configuration file on page 56

Advanced SIP settings parameters

Parameter	Default value	Description
P2342	0	Specify whether to enable Hoteling Event. Assign one of the following values to this parameter: • 0: Default • 1: No • 2: Yes
P2343	0	Specify whether to enable Call Center Status. Assign one of the following values to this parameter: • 0: Default • 1: No

Table continues…

Parameter	Default value	Description
		• 2: Yes
P2340	0	Specify whether to publish to Call Center. Assign one of the following values to this parameter: • 0: Default • 1: No • 2: Yes
P2325	0	Specify whether to enable feature key synchronization. Assign one of the following values to this parameter: • 0: Disabled • 1: Enabled
P2313	15	Specify line-seize timeout in seconds. You can assign any value in the range of 15 to 60.
P134		Specify the eventlist BLF URI.
P2318		Specify the conference URI.
P2350		Specify the URI for music-on-hold.
P1347		Specify the BLF call-pickup prefix.
P198	100	Specify the special feature. Assign one of the following values to this parameter: • 100: Standard • 101: Nortel MCS • 102: Broadsofr • 108: CBCOM • 109: RNK • 110: Sylantro • 117: Huawei IMS

Related links

Administering through the configuration file on page 56

SIP session timer settings parameters

P260	180	Specify the session expiration time in seconds. You can assign any value in the range of 90 to 64800.

Table continues…

P261	90	Specify the minimum session expiration time in seconds. This value must be lower than or equal to the value you assign to the P260 parameter. You can assign any value in the range of 90 to 64800.
P262	0	Specify whether to enable caller request timer. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P263	0	Specify whether to enable callee request timer. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P264	0	Specify whether to enable force timer. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P266	0	Specify whether to enable UAC Specify Refresher. You can assign one of the following values to this parameter: • 0: Omit • 1: UAC • 2: UAS
P267	1	Specify whether to enable UAS Specify Refresher. You can assign one of the following values to this parameter: • 1: UAC • 2: UAS
P265	0	Specify whether to enable force INVITE. You can assign one of the following values to this parameter: • 0: No • 1: Yes

Related links

Administering through the configuration file on page 56

SIP security settings parameters

Parameter	Default value	Description
P22101	Null	Specifies the trust certificate file names for TLS certificate validation. You can specify multiple file names separated with a comma. The range is up to 256 characters. E129 SIP Deskphone supports certificates in PEM format.
P2311	0	Specify whether to check the domain certificates. Assign one of the following values to this parameter: • 0: No • 1: Yes. The deskphone checks the domain certificate as defined in RFC5922.
P2306	0	Specifies whether to validate incoming messages. You can assign one of the following values to this parameter: • 0: No • 1: Yes
P258	0	Specifies whether to check SIP user ID for incoming INVITE. Assign one of the following values to this parameter: • 0: No • 1: Yes
P2347	0	Specifies whether to accept incoming SIP from proxy only. Assign one of the following values to this parameter: • 0: No • 1: Yes
P2346	0	Specifies whether to authenticate an incoming INVITE. Assign one of the following values to this parameter: • 0: No • 1: Yes

Related links

Administering through the configuration file on page 56

Audio settings parameters

Parameter	Default value	Description
P2301	0	Specifies whether to send DTMF through in-audio. Assign one of the following values: • 0: No • 1: Yes
P2302	1	Specifies whether to send DTMF through RTP. Assign one of the following values: • 0: No • 1: Yes
P2303	0	Specifies whether to send DTMF through SIP INFO. Assign one of the following values: • 0: No • 1: Yes
P79	101	Specifies the DTMF payload type. Assign value in the range of 96 to 127.
P57	0	Specifies the preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC
P58	8	Specifies the second preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC

Table continues…

Parameter	Default value	Description
P59	4	Specifies the third preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC
P60	18	Specifies the fourth preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC
P61	9	Specifies the fifth preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC
P62	98	Specifies the sixth preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC

Table continues…

Parameter	Default value	Description
P46	2	Specifies the seventh preferred vocoder. Assign any of the following values: • 0: PCMU • 2: G.726–32 • 4: G.723.1 • 8: PCMA • 9: G.722 • 18: G.729A/B • 98: iLBC
P2348	0	Specifies whether to use the first matching vocoder in 200OK SDP. Assign any of the following values: • 0: No • 1: Yes
P183	0	Specifies whether to enable SRTP mode. Assign one of the following values: • 0: Disabled • 1: Enabled but not forced • 2: Enabled and forced • 3: Optional
P291	0	Specifies whether to supports symmetric RTP. Assign any of the following values: • 0: No • 1: Yes
P50	0	Specifies whether to enable silence suppression. Assign any of the following values: • 0: No • 1: Yes. When the deskphone detects the silence, the deskphone sends a small quantity of VAD packets instead of the audio packets during the period of no talking. It is applicable only for the G.723 and G.729 codec.
P37	2	Specifies the number of voice frames transmitted per packet. It is up to 10/20/32/64 for G711/G726/G723/other codecs respectively.
P49	1	Specifies the encoding rate for G723 codec. Assign any of the following values: • 0: 6.3kbps encoding rate • 1: 5.3kbps encoding rate

Table continues…

Parameter	Default value	Description
P2323	0	Specifies the G.726-32 packing mode.Specifies the encoding rate for G723 codec. Assign any of the following values: • 0: ITU • 1: IETF
P97	1	Specifies the iLBC frame size. Assign any of the following values: • 0: 20ms • 1: 30ms
P96	97	Specifies the iLBC frame size. Assign any value in the rage of 96 to 127.
P133	1	Specifies the jitter buffer type. Assign any of the following values: • 0: Fixed • 1: Adaptive
P132	2	Specifies the jitter buffer length. Assign any of the following values: • 0:100ms • 1: 200ms • 2: 300ms • 3: 400ms • 4: 500ms • 5: 600ms • 6: 700ms • 7: 800ms

Related links

Administering through the configuration file on page 56

Call settings parameters

Parameter	Default value	Description
P29	0	Specifies whether to enable early dial. Assign any of the following values: • 0: No • 1: Yes. In this case, the SIP proxy must support must support 484 response.
P66		Specifies the dial plan prefix, that is, the prefix that the deskphone adds to the number that you dial.

Table continues…

<table>
<tr><th>Parameter</th><th>Default value</th><th>Description</th></tr>
<tr><td>P290</td><td>{ x+ | *x+ | *xx*x+ }</td><td>Specifies the dial plan for the phone. The dial plan establishes the expected number and pattern of digits for a telephone number. You can use the following rules to establish the dial plan:
• Accepted digits and characters: 0 to 9, *, #, A to D, a to d.
• x: Represents any digit from 0 to 9.
• xx+: Represents at least a 2 digit number.
• ^: Excludes the following digits or characters.
• [x-y]: Includes any digit from the range x to y.
• <x=yz>: Replaces digit x with digits y and z when dialing.
• |: Represents the Or operation.
• ,: Use comma for adding a secondary dial tone in the dial plan. Comma takes the higher precedence over Or in the dial plan. Use of caret ^ is not supported after comma. You cannot replace digits after a comma.
For example, {^1900x+ | <=1617>xxxxxxx} that specifies to block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers.</td></tr>
<tr><td>P139</td><td>20</td><td>Specifies the timeout in seconds before the phone forwards the call on no answer. Specify any value from 1 to 120.</td></tr>
<tr><td>P191</td><td>1</td><td>Specifies whether the DND, call forward, and other features can be set through the local feature code on the phone or through the ITSP feature code. Assign any of the following values:
• 0: No
• 1: Yes</td></tr>
<tr><td>P182</td><td>0</td><td>Specifies the call log settings for the phone. Assign any of the following values:
• 0: Log all calls
• 1: Log incoming and outgoing calls only. Missed calls are nor recorded.
• 2: Disable call log</td></tr>
<tr><td>P104</td><td>0</td><td>Specifies the ring tone for the account. Assign any of the following values:
• 0: System ring tone
• 1: Custom ring tone 1
• 2: Custom ring tone 2
• 3: Custom ring tone 3</td></tr>
<tr><td>P1488</td><td></td><td>Specifies the first matching rule. You can specify the matching rules through:
• A specific caller ID number, such as 8321123.</td></tr>
</table>

Table continues…

Parameter	Default value	Description
		• A defined pattern with certain length using x and + to specify a caller Id, where x is any digit from 0 to 9. For example, [345]xx specify a 3–digit number with the leading digit as 3, 4, or 5. • An Alert Info text, such as priority. The phone plays the selected ring tone if the phone receives a SIP INVITE with the Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority.
P1489	0	Specifies the ring tone that the deskphone plays when an incoming caller ID matches the first matching rule. Assign any of the following values: • 0: System ring tone • 1: Custom ring tone 1 • 2: Custom ring tone 2 • 3: Custom ring tone 3
P1490		Specifies the second matching rule. You can specify the matching rules through: • A specific caller ID number, such as 8321123. • A defined pattern with certain length using x and + to specify a caller Id, where x is any digit from 0 to 9. For example, [345]xx specify a 3–digit number with the leading digit as 3, 4, or 5. • An Alert Info text, such as priority. The phone plays the selected ring tone if the phone receives a SIP INVITE with the Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority.
P1491	0	Specifies the ring tone that the deskphone plays when an incoming caller ID matches the second matching rule. Assign any of the following values: • 0: System ring tone • 1: Custom ring tone 1 • 2: Custom ring tone 2 • 3: Custom ring tone 3
P1492		Specifies the second matching rule. You can specify the matching rules through: • A specific caller ID number, such as 8321123. • A defined pattern with certain length using x and + to specify a caller Id, where x is any digit from 0 to 9. For example, [345]xx specify a 3–digit number with the leading digit as 3, 4, or 5. • An Alert Info text, such as priority. The phone plays the selected ring tone if the phone receives a SIP INVITE with the Alert-Info header in the following format: Alert-Info: <http://127.0.0.1>; info=priority.

Table continues…

Parameter	Default value	Description
P1493	0	Specifies the ring tone that the deskphone plays when an incoming caller ID matches the second matching rule. Assign any of the following values: • 0: System ring tone • 1: Custom ring tone 1 • 2: Custom ring tone 2 • 3: Custom ring tone 3
P1328	60	Specifies the timeout in seconds for rings on no answer. You can specify any value in the range of 30 to 3600.
P65	0	Specifies whether the From header in the outgoing INVITE messages is set to anonymous. Assign any of the following values: • 0: No • 1: Yes. The deskphone blocks the caller ID.
P129	0	Specifies whether the phone rejects the anonymous calls. Assign any of the following values: • 0: No • 1: Yes
P90	0	Specifies whether the phone automatically turns on the speaker phone to answer incoming calls after a short reminding beep. Assign any of the following values: • 0: No • 1: Yes
P298	0	Specifies whether the phone automatically turns on the speaker phone to answer incoming calls after a short reminding beep based on the SIP Call-Info header that the server sends. Assign any of the following values: • 0: No • 1: Yes
P135	0	Specifies whether the Refer-To header uses the Contact header information of the transferred target for attended transfer. Assign any of the following values: • 0: No • 1: Yes
P2304	0	Specifies whether the phone transfers the call to another party if the conference initiator hangs up. Assign any of the following values: • 0: No • 1: Yes
P85	4	Specifies the timeout in seconds when the phone dials the digits if no key is pressed after the timeout. Specify any value in the range of 1 to 15.

Table continues…

Parameter	Default value	Description
P72	1	Specifies whether the # key is used to immediately dial out the input digits. Assign any of the following values: • 0: No • 1: Yes

Related links

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Basic network settings parameters

Parameter	Default value	Description
P1415	0	Specifies Internet Protocol that the deskphone uses. Assign any of the following values: • 0: IPv4 • 1: IPv6
P8	0	Specifies the method for the phone to obtain the IPv4 address. Assign any of the following values: • 0: DHCP. You must also specify the parameters P146 and P148. • 1: Static. You must also specify the parameters P9, P10, P11, P12, P13, P14, P15, P16, P17, P18, P19, P20, P21, P22, P23, P24, P25, P26, P27, P28, P92, P93, P94, P95, P1419, P1424, P1425, P1423, P1426, P1420, P1421, and P1422. • 2: PPPoE. You must also specify the parameters P82, P83, and P269.
P146		Specifies the name of the client. This field is optional but some Internet Service Providers might require it. The maximum length is 64 characters.
P148		Specifies vendor class ID that client and servers exchange. The maximum length is 64 characters.
P82		Specifies the PPPoE account ID.
P83		Specifies the PPPoE password. Must not contain an apostrophe (‘) or a quote (“).
P269		Specifies the PPPoE service name.
P9		Specifies the first octet of the IP address. Assign any value in the range of 0 to 255.
P10		Specifies the second octet of the IP address. Assign any value in the range of 0 to 255.
P11		Specifies the third octet of the IP address. Assign any value in the range of 0 to 255.

Table continues…

Parameter	Default value	Description
P12		Specifies the fourth octet of the IP address. Assign any value in the range of 0 to 255.
P13		Specifies the first octet of the subnet mask. Assign any value in the range of 0 to 255.
P14		Specifies the second octet of the subnet mask. Assign any value in the range of 0 to 255.
P15		Specifies the third octet of the subnet mask. Assign any value in the range of 0 to 255.
P16		Specifies the fourth octet of the subnet mask. Assign any value in the range of 0 to 255.
P17		Specifies the first octet of the default gateway. Assign any value in the range of 0 to 255.
P18		Specifies the second octet of the default gateway. Assign any value in the range of 0 to 255.
P19		Specifies the third octet of the default gateway. Assign any value in the range of 0 to 255.
P20		Specifies the fourth octet of the default gateway. Assign any value in the range of 0 to 255.
P21		Specifies the first octet of the first DNS server. Assign any value in the range of 0 to 255.
P22		Specifies the second octet of the first DNS server. Assign any value in the range of 0 to 255.
P23		Specifies the third octet of the first DNS server. Assign any value in the range of 0 to 255.
P24		Specifies the fourth octet of the first DNS server. Assign any value in the range of 0 to 255.
P25		Specifies the first octet of the second DNS server. Assign any value in the range of 0 to 255.
P26		Specifies the second octet of the second DNS server. Assign any value in the range of 0 to 255.
P27		Specifies the third octet of the second DNS server. Assign any value in the range of 0 to 255.
P28		Specifies the fourth octet of the second DNS server. Assign any value in the range of 0 to 255.
P92		Specifies the first octet of the preferred DNS server. Assign any value in the range of 0 to 255.
P93		Specifies the second octet of the preferred DNS server. Assign any value in the range of 0 to 255.
P94		Specifies the third octet of the preferred DNS server. Assign any value in the range of 0 to 255.

Table continues…

Parameter	Default value	Description
P95		Specifies the fourth octet of the preferred DNS server. Assign any value in the range of 0 to 255.
P1419	0	Specifies the method for the phone to obtain the IPv4 address. Assign any of the following values: • 0: Auto-configured • 1: Statically configured:
P1424		Specifies the first DNS Server for IPv6 address.
P1425		Specifies the second DNS Server for IPv6 address.
P1423		Specifies the preferred DNS Server for IPv6 address.
P1426		Specifies the IPv6 address type. Assign any of the following values: • 0: Full static. You must also specify the parameters P1420 and P1421. • 1: Prefix static. You must also specify the parameter P1422
P1420		Specifies the static IPv6 address.
P1421		Specifies the IPv6 prefix length.
P1422		Specifies the IPv6 prefix.

Related links

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Advanced network settings parameters

Parameter	Default value	Description
P7901	0	Specifies whether to enable the 802.1X mode. Assign any of the following values: • 0: Disable • 1: EAP-MD5
P7902		Specifies the identity for the 802.1X mode. The maximum length is 64 characters.
P7903		Specifies the MD5 password. The maximum length is 64 characters.
P1552		Specifies the HTTP proxy URL where the deskphone sends the packets. The proxy server is an intermediary to route the packets to the destination.
P1553		Specifies the HTTPS proxy URL where the dseskphone sends the packets. The proxy server is an intermediary to route the packets to the destination.
P38	12	Specifies the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv, or MPLS. Assign any value in the range of 0 to 63.

Table continues…

<table>
<tr><th>Parameter</th><th>Default value</th><th>Description</th></tr>
<tr><td>P51</td><td>0</td><td>Specifies the VLAN tag of the Layer 2 QoS packets. Assign any value in the range of 0 to 4094.</td></tr>
<tr><td>P87</td><td>0</td><td>Specifies the priority value of the Layer 2 QoS packets. Assign any value in the range of 0 to 7.</td></tr>
<tr><td>P1348</td><td>0</td><td>Specifies the PC port mode. Assign any of the following values:
• 0: Enable
• 1: Disabled
• 2: Mirrored
If you assign the value 2, the traffic in the LAN port goes through the personal computer port and the system can capture packets if you connect the computer to the PC port.</td></tr>
<tr><td>P1684</td><td>1</td><td>Specifies whether to enable the LLDP service. Assign any of the following values.
• 0: Disabled
• 1: Enabled
LLDP-MED is supported only for VLAN priority and Tag value. If Avaya-proprietary LLDP is not supported, use DHCP to receive file server or SIP server address parameters.</td></tr>
</table>

Related links

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Web access parameters

Parameter	Default value	Description
P196	-	Specifies user password.
P2	-	Specifies administrator password.

Related links

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Deskphone upgrading and provisioning parameters

During the upgrade, if the deskphone receives a call, the deskphone starts ringing. If the user does not answer the call, the deskphone continues to ring till the upgrade completes. Therefore, you must plan the upgrade during the off hours.

Parameter	Default value	Description
P238		Specifies whether the deskphone checks and installs a new firmware every time the deskphone boots up. Assign any of the following values: • 0: Always check for new firmware • 1: Check new firmware only when the pre or suffix of the firmware changes 2: Always skip the firmware check
P1359		Specifies the password that the deskphone uses to decrypt the encrypted XML configuration file.
P1360		Specifies the user name to login to the HTTP and HTTPS server.
P1361		Specifies the password to login to the HTTP and HTTPS server.
P212	1	Specifies the firmware upgrade method. Assign any of the following values: • 0: TFTP • 1: HTTP • 2: HTTPS **Note:** If you select HTTPS, you must install CA certificates before the deskphone upgrade. For more information about installing the CA certificates, see *Installing and Maintaining Avaya E129 SIP Deskphone*.
P192		Specifies the server path for the firmware.
P237		Specifies the address of the configuration file server.
P232		Specifies the prefix for the firmware file. The deskphone downloads the firmware file only if the encrypted prefix matches the one that you specified.
P233		Specifies the postfix for the firmware file. The deskphone downloads the configuration file only if the encrypted postfix matches the one that you specified.
P234		Specifies the prefix for the configuration file. The deskphone downloads the configuration file only if the encrypted prefix matches the one that you specified.
P235		Specifies the postfix for configuration file. The deskphone downloads the configuration file only if the encrypted postfix matches the one that you specified.
P145	1	Specifies whether the DHCP option 43 and option 66 override the SIP server. If you enable the DHCP option 66, the system can redirect the TFTP server. Assign any of the following values: • 0: No • 1: Yes

Table continues…

Parameter	Default value	Description
P1411	0	Specifies whether the DHCP option 120 overrides the SIP server. Assign any of the following values: • 0: No • 1: Yes
P194	0	Specifies whether the deskphone checks and installs a new firmware. Assign any of the following values: • 0: No • 1: Yes
P193	10080	Specifies time in minutes when the deskphone checks the HTTP/TFTP server for a firmware upgrade or configuration file changes. You can assign any value in the range of 60 to 86400.
P285	1	Specifies the hour of the day when the deskphone checks the HTTP/TFTP server for a firmware upgrade or configuration file changes. You can assign any value in the range of 0 to 23.
P286	1	Specifies the day of the week when the deskphone checks the HTTP/TFTP server for a firmware upgrade or configuration file changes. You can assign any value in the range of 0 to 6.
P240	0	Specifies whether the deskphone authenticates the configuration file before accepting the file. Assign any of the following values: • 0: No • 1: Yes

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Syslog parameters

Parameter	Default value	Description
P207		Specifies the URL or IP address of the syslog server. The maximum length is 64 characters.
P208	0	Specifies the level of syslog logging. Assign any of the following values: • 0: None • 1: Debug • 2: Info • 3: Warning • 4: Error

Table continues…

Parameter	Default value	Description
P1387	0	Specifies whether the system includes the SIP log in the syslog messages. Assign any of the following values: • 0: No • 1: Yes
P1438	1	Specifies whether the phone autorecovers when it does not run normally. Assign any of the following values: • 0: No • 1: Yes

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Language parameters

Parameter	Description	Description
P1362	auto	Specifies the display language on the phone. Assign any of the following values: • ar : Arabic • cz: Czech • de: Deutsh • en: English • es: Spanish • fr: Francais • he: Hebrew • hr: Hrvatski • hu: Magyar • it: Italiano • nl: Dutch • pl: Polski • pt: Portugue • ru: Russian • sl: Slovenian • tr: Turkish • zh-tw: Traditional chinese

Table continues…

Parameter	Description	Description
		• zh: Simplified chinese • auto: Automatic • gxp: Downloaded language
P399		Specifies the language file postfix for the language that the phone downwloads.

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Security settings parameters

Parameter	Default value	Description
P1357	0	Specifies the access control for keypad Menu settings. Assign any of the following values: • 0: Unrestricted. Displays all menu options on the deskphone. • 1: Basic settings only. Hides the Config menu option on the deskphone. • 2: Constraint mode. Hides the Config and Factory Functions menu options on the deskphone.
P1382	1	Specifies whether holding the star (*) key for 4 seconds locks the keypad. Assign any of the following values • 0: No • 1: Yes
P1383		Specifies the password to lock and unlock the keypad.
P280		Specifies the identity certificate file for the phone to authenticate over the SIP server through TLS. Supports the PEM file format only.
P279		Specifies the private key for the SIP TLS identity certificate in PEM format.
P281		Specifies the passphrase of the private key for the SIP TLS identity certificate in text format.
P1650	1	Specifies the protocol for the Web interface. Assign any of the following values: • 0: HTTPS • 1: HTTP
P276	0	Specifies whether to disable SSH access. Assign any of the following values: • 0: No • 1: Yes

Related links

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XML phone book configuration parameters

Parameter	Default value	Description
P330	0	Specifies whether to enable the XML phone book download. Select one of the following options: • 0: Disabled • 1: Enabled, HTTP • 2: Enabled, TFTP • 3: Enabled, HTTPS
P331		Specifies the server path, IP address or URL, to download the XML phone book XML. You can enter up to 256 characters. Use one of the following syntax to provide the server path: • IP_address[:port]/dir • Hostname[:port]/dir For example, 192.168.40.10/XMLphone bookdir service.mycompany.com/XML/phone book If you do not specify the port, the system uses port 80 as a default port for HTTP, port 443 as a default port for HTTPS, and port 69 as a default port for TFTP. Do not specify the phone book file name in the server path, but only the directory path as Avaya E129 SIP Deskphone looks for the `(null)_phonebook.xml` file to download the phone book. If you enter the server path in the browser, the browser displays the XML phone book. You can save the XML phone book on your personal computer.
P332	0	Specifies the phone book download interval in minutes. 0 disables the automatic download. You can specify any value in the range of 5 to 720.
P333	1	Specifies whether the phone deletes the manual entries when the phone downloads the XML phone book. Assign any of the following values: • 0: No • 1: Yes

Table continues…

Parameter	Default value	Description
P1526		Specifies the **Phonebook** key behavior. Assign one of the following values: • 0: Default • 1: LDAP search • 2: Local phone book • 3: Local group

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LDAP settings parameter

Parameter	Default value	Description
P8020		Specifies the IP address or DNS name of the LDAP server.
P8021	389	Specifies the port of the LDAP server. Assign any value in the range of 1 to 65535.
P8022		Specifies the LDAP search base. The search base is location in the directory from where the search begins. For example, dc=grandstream, dc=com.
P8023		Specifies the bind user name for querying the LDAP server. If the LDAP server allows anonymous binds, you can leave this parameter blank.
P8024		Specifies the bind password for querying the LDAP server. If the LDAP server allows anonymous binds, you can leave this parameter blank.
P8025		Specifies the filter that the phone uses for number lookups. For example, (|(telephoneNumber=%)(Mobile=%) returns all records which has the telephone number field or mobile field starting with the prefix that you enter.
P8026		Specifies the filter that the phone uses for name lookups. For example, (|(cn=%)(sn=%)) returns all records which has the cn field or sn field starting with the prefix that you enter.
P8027		Specifies the protocol version for the phone to send the bind requests.
P8028		Specifies the space-separated name attributes that the phone returns for an LDAP search. For example, cn sn description.
P8029		Specifies the space-separated number attributes that the phone returns for an LDAP search. For example, telephoneNumber Mobile.
P8030		Specifies space-separated attributes that the phone displays. You can display up to 3 attributes. For example, %cn %sn %telephoneNumber.
P8031	50	Specifies the maximum number of results that an LDAP search returns. You can specify any value in the range of 0 to 32000.

Table continues…

Parameter	Default value	Description
P8032	30	Specifies the interval in seconds for the server to process the request and for the client to wait for the sever result. You can specify any value in the range of 0 to 180.
P8033	0	Specifies whether the phone sorts LDAP search result. Assign any of the following values: • 0: No • 1: Yes
P8035	0	Specifies whether to enable number searching at the time of dialing and receiving calls. Assign any of the following values: • 0: No • 1: Yes
P8036	0	Specifies the space-separated display name for the attributes that an LDAP search looks up for incoming and outgoing calls. Assign any of the following values: • 0: No • 1: Yes

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General settings parameters

Name	Default value	Description
P39	5004	Specifies the local RTP port that the deskphone uses to listen and transmit. The value must be an even number and in the range of 1024 to 65400.
P78	1	Specifies whether the deskphone forces a random generation of both the local SIP and RTP ports. Assign any of the following values: • 0: No • 1: Yes
P2907	1	Specifies whether to enable the auto login feature. Assign any of the following values: • 0: No • 1: Yes

Related links

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Call features settings parameters

Parameter	Default value	Description
P71		Specifies the number that the deskphone dials when the deskphone is off-hook.
P1485	30	Specifies the time in seconds after which the deskphone goes on-hook after being off-hook. You can specify any value in the range of 10 to 60.
P91	0	Specifies whether to disable the call waiting feature. Assign any of the following values: • 0: No • 1: Yes
P186	0	Specifies whether to disable the call waiting tone when the Call Waiting feature is active. Assign any of the following values: • 0: No • 1: Yes
P184	0	Specifies whether the user can dial an IP address under the same LAN or VPN segment by entering the last octet in the IP address. Assign any of the following values: • 0: No • 1: Yes
P338	0	Specifies whether the deskphone displays the DTMF digits that the user enters during a call. Assign any of the following values: • 0: No • 1: Yes
P1565	0	Specifies whether to enable the Idle Mute feature. Assign any of the following values: • 0: No • 1: Yes
P1525		Specifies the number that the deskphone dials as DTMF when the user presses the Transfer button
P1376	0	Specifies whether the deskphone uses the attended transfer by default. Assign any of the following values: • 0: No • 1: Yes **Note:** Only blind transfer is supported if both the lines are occupied.

Table continues…

Parameter	Default value	Description
P1406	0	Specifies whether the deskphone replaces # by %23 for some special situations. Assign any of the following values: • 0: No • 1: Yes
P1561	0	Specifies whether to enable the Click-To-Dial feature. Assign any of the following values: • 0: No • 1: Yes
P1696	0	Specifies whether the deskphone blinks the Message LED for an incoming call. Assign any of the following values: • 0: No • 1: Yes
P1433	300	Specifies the interval in seconds for the deskphone to save the call history to the flash memory. 0 implies that the option is disabled.
P1434	200	Specifies the number of unsaved logs before the deskphone writes them to the flash memory. Assign any value in the rage of 0 to 500.
P1311	0	Specifies whether local conferences are enabled. • 0: Disabled • 1: Enabled

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Ring tone parameters

Parameter	Default value	Decsription
P345	Null	Specifies frequency and cadence for system ring tone. Use the following syntax to specify frequencies and cadence for a ring tone. f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; where • f1 and f2 are frequencies in Hz. • c is cadence on and off in ms. The system supports up to three cadences. For example, f1=440,f2=480,c=200/400;

Table continues…

Parameter	Default value	Decsription
P343	Null	Specifies frequency and cadence for dial tone. Use the syntax as specified in the parameter P345.
P344	Null	Specifies frequency and cadence for message waiting. Use the syntax as specified in the parameter P345.
P346	Null	Specifies frequency and cadence for ring back tone. Use the syntax as specified in the parameter P345.
P347	Null	Specifies frequency and cadence for ring back tone. Use the syntax as specified in the parameter P345.
P1555	Null	Specifies frequency and cadence for call waiting tone gain. Use the syntax as specified in the parameter P345.
P348	Null	Specifies frequency and cadence for busy tone. Use the syntax as specified in the parameter P345.
P349	Null	Specifies frequency and cadence for reorder tone. Use the syntax as specified in the parameter P345.

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Audio control parameters

Parameter	Default value	Description
P1312	0	Specifies the mode for the headset button when a headset is connected to the deskphone. Assign any of the following values: • 0: Default mode 1: Toggle Headset/Speaker
P1487	0	Specifies the headset type. Assign any of the following values: • 0: Normal, an RJ9 headset • 1: Plantronics EHS
P1439	0	Specifies whether to enable the speaker to ring when the deskphone uses the headset in the Toggle Headset/Speaker mode. Assign any of the following values: • 0: No • 1: Yes
P1301	0	Specifies the transmission gain of the headset. Assign any of the following values: • 0: 0 decibel • 1: –6 decibel

Table continues…

Parameter	Default value	Description
		• 2: +6 decibel
P1302	0	Specifies the receiving gain of the headset. Assign any of the following values: • 0: 0 decibel • 1: –6 decibel • 2: +6 decibel
P1464		Specifies the transmission gain of the handset. Assign any of the following values: • 0: 0 decibel • 1: –6 decibel • 2: +6 decibel

Related links

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LCD display parameter

Parameter	Default value	Description
P1329	10	Specifies the LCD contrast level. Specify any value in the range of 0 to 20.

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Date and time parameters

Parameter	Default value	Description
P30		Specifies the URL or IP address of the NTP server. The phone can obtain the date and time from the server.
P144	1	Specifies whether the DHCP Option 42 overrides the NTP server. Assign any one of the following values: • 0: No

Table continues…

Parameter	Default value	Description
		• 1: Yes. Override the NTP server that you configured.
P64		Specifies the time zone based on which the phone displays the date and time.
P143	1	Specifies whether DHCP Option 2 overrides the time zone settings. Assign any one of the following values: • 0: No • 1: Yes. Overrides the Time Zone setting if available.
P246		Specifies the user-defined time zone. Use the following syntax to define the time zone: std offset dst [offset], start [/time], end [/time] The default is set to MTZ+6MDT +5,M4.1.0,M11.1.0 , where • MTZ+6MDT+5 indicates a time zone with six hours offset with one hour ahead of the U.S central time. Plus (+) indiactes the local time zone is west of the Prime Meridian. Minus (-) indicates the local time zone is east of the Prime Meridian. • M4.1.0,M11.1.0 in which the first number indicates the month, where 1 corresponds to January, 2 corresponds to February, and so on. The second number indicates the nth iteration of the weekday, where 1 corresponds to Sunday, 2 corresponds to Monday, and so on. The third number indicates weekday, where 0 corresponds to Sunday, 1 corresponds to Monday, and so on.

Table continues…

Parameter	Default value	Description
P102		Specifies the date display format on the LCD. Assign any of the following values: • 0: yyyy-mm-dd • 1: mm-dd-yyyy • 2: dd-mm-yyyy • 3: dddd, MMMM dd • 4: MMMM dd, dddd
P122		Specifies the time display format on the LCD. Assign any of the following values: • 0: 12 hour format • 1: 24 hour format

Related links

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XML applications settings parameters

Parameter	Default value	Description
P340	0	Specifies whether to download the idle screen XML through HTTP, HTTPS, or TFTP. Assign any of the following values: • 0: Disabled • 1: Enabled, HTTP • 2: Enabled, TFTP • 3: Enabled, HTTPS
P1349	0	Specifies whether the deskphone downloads the idle screen XML at the time of starting up. Assign any one of the following values: • 0: No • 1: Yes
P1343		Specifies the name of the idle screen XML file that the

Table continues…

Parameter	Default value	Description
		deskphone downloads. Assign any one of the following values: • 0: No • 1: Yes
P341		Specifies the server path in the host/path format from where the deskphone downloads the idle screen XML file. You can specify an IP address or a URL with up to 256 characters.

Related links

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Configuring speed dial

About this task

Speed dial enables you to provide a one-touch direct dial softkey on the idle screen by modifying the Idle Screen XML file.

Before you begin

Ensure that you have the access to the Idle Screen XML file.

Procedure

Add an element in the Idle Screen XML with the following syntax:

```
<Softkeys>

<SoftKeyaction=”Dial" label="SpeedDial" commandId=”0"
commandArgs=”1002"/>

<Softkeys>
```

where:

- <label> defines the softkey display name on the LCD. A maximum of 8 characters is displayed on the idle screen softkey.
- <commandId> specifies the account index to dial out from. Value is always 0.
- <commandArgs> specifics the phone number to dial

Related links

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Sample configuration file

Following are the samples of configuration file in the text and XML formats.

Configuration file in the text format

```
##########################################################################################
######
## Configuration Template For Avaya E129 Firmware Version
1.0.5.32                               ##
##########################################################################################
######
####################################################################
# Account Settings                                                ##
####################################################################
####################################################################
# Account 1                                                       ##
####################################################################
#-----------------------------------------
# Account 1/General Settings
#-----------------------------------------
# Account Active. 0 - No, 1 - Yes. Default is 1
# Number: 0, 1
# Mandatory
P271 = 1

# Account Name
# String
# P270 =

# SIP Server
# String
P47 =

# Secondary SIP Server
# String
P2312 =

# Display Name
# String
P3 =

# Voice Mail UserID
# String
P33 =

#-----------------------------------------
# Account 1/Network Settings
#-----------------------------------------
# DNS Mode. 0 - A Record, 1 - SRV, 2 - NAPTR/SRV, 3 - Use Configured IP. Default is 0
# Number: 0, 1, 2, 3
# Mandatory
P103 = 0

# Primary IP. Maximum 15 characters
# String
P2308 =

# Backup IP 1
# String
P2309 =

# Backup IP 2
```

```
# String
P2310 =

# Proxy-Require (A SIP extension to enable firewall penetration). Max length is 64
characters
# String
P197 =

...
...
...


###############################################################################
##  Settings/XML Applications
###############################################################################
# Enable Idle Screen XML Download
# 0 - Disabled, 1 - Enabled, HTTP, 2 - Enabled, TFTP, 3 - Enabled, HTTPS. Default is 0
# Number: 0, 1, 2, 3
# Mandatory
P340 = 0

# Download Screen XML At Boot-up. 0 - No, 1 - Yes. Default is 0
# Number: 0, 1
# Mandatory
P1349 = 0

# Use Custom File Name. 0 - No, 1 - Yes. Default is 0
# Number: 0, 1
# Mandatory
P1343 = 0

# Disable Conference. 0 - No, 1 - Yes. Default is 0
# Number: 0, 1
# Mandatory
P1311 = 0

# Idle Screen XML Server Path
# This is a string of up to 256 characters that should contain a path to the XML file
# It MUST be in the host/path format. For example: directory.grandstream.com/engineering
# String
P341 =
```

Configuration file in the XML format

```
<?xml version="1.0" encoding="UTF-8" ?>
<gs_provision version="1">
<!-- <mac></mac> -->
<config version="1">

<!-- Generic Configuration Template For E129 Firmware Version 1.0.5.58  -->
<!--  Configuration File Version 0.5 -->

<!--Account Active. 0 - No, 1 - Yes. Default is 1 -->
<!-- # Number: 0, 1 -->
<!-- # Mandatory -->
<!-- <P271>1</P271> -->

<!-- Account Name -->
<!-- # String -->
<!--<P270></P270>-->

<!-- SIP Server-->
<!-- # String -->
<!--<P47></P47>-->
```

```
<!-- Secondary SIP Server -->
<!-- # String -->
<!--<P2312></P2312>-->

<!-- Display Name -->
<!-- # String -->
<!--<P3></P3>-->

<!-- Voice Mail UserID -->
<!-- # String -->
<!--<P33></P33>-->


<!-- ########################################## -->
<!-- # Account 1/Network Settings -->
<!-- ########################################## -->
<!-- # DNS Mode. 0 - A Record, 1 - SRV, 2 - NAPTR/SRV, 3 - Use Configured IP. Default is
0 -->
<!-- # Number: 0, 1, 2, 3 -->
<!-- # Mandatory -->
<!-- <P103>0</P103> -->

<!-- # Primary IP. Maximum 15 characters -->
<!-- # String -->
<!--<P2308></P2308>-->

<!-- # Backup IP 1 -->
<!-- # String -->
<!--<P2309></P2309>-->

<!-- # Backup IP 2 -->
<!-- # String -->
<!--<P2310></P2310>-->

<!-- # Proxy-Require (A SIP extension to enable firewall penetration). Max length is 64
characters -->
<!-- # String -->
<!--<P197></P197>-->

...
...
...

<!-- ############################################################################# -->
<!-- ##  Settings/XML Applications -->
<!-- ############################################################################# -->
<!-- # Enable Idle Screen XML Download -->
<!-- # 0 - Disabled, 1 - Enabled, HTTP, 2 - Enabled, TFTP, 3 - Enabled, HTTPS. Default is
0 -->
<!-- # Number: 0, 1, 2, 3 -->
<!-- # Mandatory -->
<!-- <P340>0</P340> -->

<!-- # Download Screen XML At Boot-up. 0 - No, 1 - Yes. Default is 0 -->
<!-- # Number: 0, 1 -->
<!-- # Mandatory -->
<!-- <P1349>0</P1349> -->

<!-- # Use Custom File Name. 0 - No, 1 - Yes. Default is 0 -->
<!-- # Number: 0, 1 -->
<!-- # Mandatory -->
<!-- <P1343>0</P1343> -->

<!-- # Disable Conference. 0 - No, 1 - Yes. Default is 0 -->
```

```
<!-- # Number: 0, 1 -->
<!-- # Mandatory -->
<!-- <P1311>0</P1311> -->

<!-- # Idle Screen XML Server Path -->
<!-- # This is a string of up to 256 characters that should contain a path to the XML
file -->
<!-- # It MUST be in the host/path format. For example: directory.company.com/engineering
-->
<!-- # String -->
<!-- <P341></P341> -->
</config>
</gs_provision>
```

Related links

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