vtech

ErisTerminal[®] SIP Deskset VSP861

Administrator and Provisioning Manual





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PREFACE

Congratulations on your purchase of this VTech product. Please thoroughly read this manual for all the feature operations and troubleshooting information necessary to install and operate your new VTech product. You can also visit our website at businessphones.vtech.com or call 1 (888) 370-2006.

This administrator and provisioning manual contains detailed instructions for installing and configuring your VSP861 SIP deskset with software version 2.0.4.0. See "Using the Status menu" on page 30 for instructions on checking the software version on the VSP861. Please read this manual before installing the product.

Please print this page and record the following information regarding your product:

| r loade print the page and recert the renewing information regarding your product. |
|---|
| Model number: VSP861 |
| Type: Small to medium business SIP-endpoint deskset |
| Serial number: |
| Purchase date: |
| Place of purchase: |
| Both the model and serial numbers of your VTech product can be found on the bottom of the console. |
| Save your sales receipt and original packaging in case it is necessary to return your telephone for warranty service. |

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Text Conventions

Table 1 lists text formats and describes how they are used in this guide.

Table 1. Description of Text Conventions

| Text Format | Description |
|---|--|
| Screen | Identifies text that appears on a device screen or a WebUI page in a title, menu, or prompt. |
| HARD KEY or DIAL-PAD KEY | Identifies a hard key, including the dial-pad keys. |
| Notes provide important information about a feature or procedure. | Example of a Note. |
| A caution means that loss of data or caution unintended circumstances may result. | Example of a Caution. |

Audience

This guide is written for installers and system administrators. It assumes that you are familiar with networks and VoIP, both in theory and in practice. This guide also assumes that you have ordered your IP PBX equipment or service and selected which PBX features you want to implement. This guide references specific IP PBX equipment or services only for features or settings that have been designed for a specific service. Please consult your equipment supplier or service provider for recommended switches, routers, and firewall and NAT traversal settings, and so on.

As the VSP861 SIP deskset becomes certified for IP PBX equipment or services, VTech may publish interop guides for those specific services. The interop guides will recommend second-party devices and settings, along with VSP861-specific configurations for optimal performance with those services. For the latest updates, visit our website at <a href="https://doi.org/10.1007/bit.100

Related Documents

The **VSP861 Quick Start Guide** contains a quick reference guide to the VSP861 external features and brief instructions on connecting the VSP861 to a working IP PBX system.

The **VSP861 User Guide** contains a quick reference guide, full installation instructions, instructions for making and receiving calls, and a guide to all user-configurable settings.

The documents are available from our website at *businessphones.vtech.com*.

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CHAPTER 1

INTRODUCING THE VSP861

This administrator and provisioning guide contains detailed instructions for configuring the VSP861 SIP deskset. Please read this guide before attempting to configure the VSP861.

Some of the configuration tasks described in this chapter are duplicated in the Web User Interface (WebUI) described in the next chapter, but if you need to assign static IP addresses, they must be set at each device.

This chapter covers:

- "About the VSP861 deskset" on page 10
- "Quick Reference Guide" on page 11
- "Programmable Feature Keys" on page 15
- "Network Requirements" on page 19
- "VSP861 Configuration Methods" on page 20
- "Adding a Custom Logo" on page 21
- "Using an SD card" on page 22.



About the VSP861 deskset

The VTech VSP861 SIP deskset is a business phone designed to work with popular SIP telephone (IP PBX) equipment and services. Once you have ordered and configured your SIP equipment or service, the VSP861 enables you to make and receive calls as you would with any other business phone. The VSP861 provides calling features like hold, transfer, conferencing, speakerphone, speed-dial numbers and one-touch directory access.

The VSP861 deskset features include:

- Speakerphone, headset, hold and mute
- Up to 8 SIP account registrations
- Up to 10 active SIP sessions
- 3-way conferencing
- 30 programmable feature keys
- Message Waiting alert LED
- Dual GigE Ethernet ports
- Power over Ethernet enabled
- DECT cordless headset and cordless handset support
 For information about registering a cordless headset, see the VSP861 Deskset
 User's Guide. For information about registering a cordless handset, see the
 VSP601 Handset Quick Start Guide.
- Micro SD card slot for call recording, network traces, system logging, firmware updates and configuration file backup and updates.
- 1000-entry Call Log

There are two network ports, known as the Ethernet port and PC port, at the back of the VSP861. The Ethernet port allows the VSP861 deskset to connect to the IP PBX. The PC port is for another device such as a personal computer to connect to the Ethernet network through the VSP861.

You can configure the VSP861 using the menus on the phone, a browser-based interface called the WebUI, or an automatic provisioning process (see "Provisioning Using Configuration Files" on page 132). The WebUI enables you to configure the VSP861 using a computer that is connected to the same Local Area Network. The WebUI resides on the VSP861, and may get updated with firmware updates.

The VSP861 SIP deskset supports intercom and call transfers between system extensions and can connect you and two other parties on the same conference call.

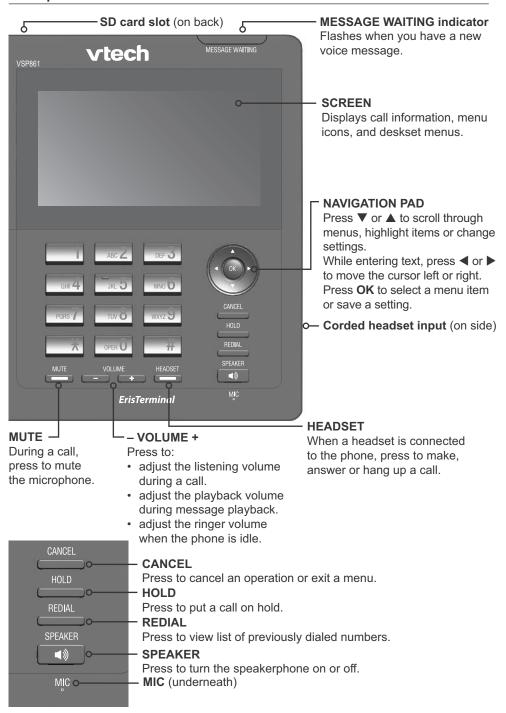
The VSP861 has 30 programmable feature keys. You can program these keys for quick dial, busy lamp field, line access or any of the functions described in "Programmable Feature Keys" on page 15.



Quick Reference Guide

The controls you will need to use to configure the VSP861 manually are described below.

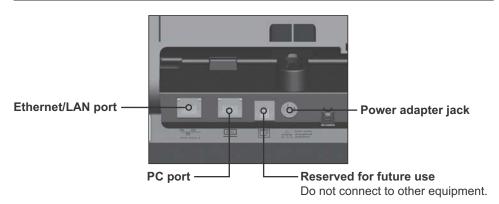
Front panel features





The external features that are relevant to installation and configuration are described below.

Rear panel connection ports





Home screen

The Home screen appears after the phone is connected and configured for your SIP PBX service. The Home screen indicates that you have no active or held calls.

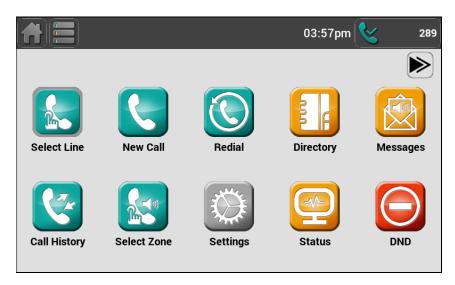
If you have an active call or are viewing a menu, you can view the Home screen at any time by pressing

Pressing a shortcut on the Home screen opens the application associated with that shortcut.

To view additional pages of shortcuts, press or less o



You can program and reconfigure these keys to suit your needs. See "Configuring the Home screen shortcuts" on page 27.



The Home screen is one of three possible default screens. The default screen appears when your phone is idle. You can select your prefered default screen on the App Settings menu. See "Configuring the Default screen" on page 25.

Your system administrator can add a custom background to the Home screen. The appearance of your Home screen and the available shortcuts may differ from that shown.



Feature Keys

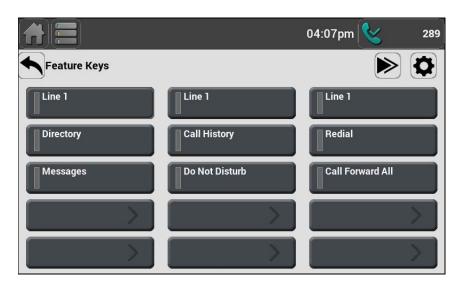
The deskset offers 30 virtual feature keys that give you quick access to phone features and call functions. You can program and reconfigure these keys to suit your needs. See "Programmable Feature Keys" on page 15 and "Configuring the Feature keys" on page 28.

There are 15 feature keys displayed per page.

To view additional pages of feature keys, press or large or large



If you have an active call or are viewing a menu, you can view the Feature Keys screen at any time by pressing



A multicolored indictor on each feature key shows you call status and other information. See "LED Behavior" on page 18.



Programmable Feature Keys

The table below lists the default settings for the programmable feature keys (PFKs). The key assignments on your phone may be different. Some keys may be programmed as Quick Dial keys, for example. You can assign functions to programmable keys using the

phone menu (Settings > User Settings > Programmable Keys > Feature Keys), using the WebUI, or via provisioning and the configuration file.

For more information about assigning functions to programmable keys using the phone menu, see "Configuring the Feature keys" on page 28 and the User Guide.

To assign functions to programmable keys using the WebUI, see "Programmable Feature Keys" on page 72.

For the programmable key configuration file parameters, see "pfk Module: Programmable Feature Key Settings" on page 192.

Table 2. Programmable key default settings

| Key Number | Setting |
|------------|------------------|
| 1 | Line—Account 1* |
| 2 | Line—Account 1 |
| 3 | Line—Account 1 |
| 4 | Directory |
| 5 | Call History |
| 6 | Redial |
| 7 | Messages |
| 8 | Do Not Disturb |
| 9 | Call Forward All |
| 10 | None |
| 11 | None |
| 12 | None |
| 13– | None |

^{*} You can assign more than one key to an account. For example, you can configure Line keys 1 and 2 to access Account 1, and Line keys 3 and 4 to access Account 2. Use the key label card to label the keys appropriately for VSP861 users after configuration.



Other possible programmable key settings

- Do Not Disturb All—Turns DND on for all registered accounts.
- Call Forward No Answer—Turns Call Forward No Answer (CFNA) on and off.
 When CFNA is on, unanswered incoming calls are forwarded to another number after a specified delay.
- Call Forward Busy—Turns Call Forward Busy (CFB) on and off. When CFB is on, incoming calls are forwarded to another number when the line is busy.
- Quick Dial
- BLF—Busy lamp field keys let you monitor activity at other phones. The key LED indicates call status.
- Page—If this feature is enabled, press the Page key to call one or a group of phones. You can configure pages to be automatically answered. See "SIP Account Management" on page 51.
- Multicast page—Press the Multicast page key to page all phones in a pre-defined paging zone. See "Paging Zones" on page 79.
- Park Call—Dials the access code to park your current call. To program access codes, see "SIP Account Management" on page 51.
- Retrieve Parked Call—Dials the access code to retrieve a parked call.
- In Call DTMF—Dials a string of numbers while you are on a call. For example, pressing the key might dial a conference access code.
- Call Return—Dials the number of the last missed call.
- Group Call Pickup—Dials the Group Call Pickup code, allowing you to answer a call ringing at any extension within an admin-defined group.
- Direct Call Pickup—Dials the Direct Call Pickup code, allowing you to answer a call ringing at a specific extension. After pressing the button, you may need to enter the extension number manually.
- Prefix Dialing—Automatically goes off-hook and dials a prefix (hidden from the user).
- Call Handling Profile—For Comverse server-side feature contol. Dials the call handling profile code along with the profile index number to activate the profile.
- Hunt Group—Dials the hunt group extension number.
- Secretarial Filtering—Turns Comverse secretarial filtering on or off.
- Phone Lock—Locks the phone or enables the user to unlock the phone with a PIN. See the phone lock section.
- Flash—Performs a hook flash, as defined in "Programmable Feature Keys" on page 72.



XML App—Displays a user-initiated (pulled) XML-based screen or menu.



LED Behavior

The programmable feature keys have LEDs that indicate various states.

Table 3. VSP861 LED behavior

| Key function | LED Activity | Description |
|--------------------------|--|---|
| Account | Flashing ORANGE Steady GREEN Quickly flashing GREEN Slowly flashing GREEN | Account not registered Dialing or on a call Ringing incoming call Held call |
| Shared account | Steady ORANGE Slowly flashing ORANGE | Shared account is on a call Shared account is on hold |
| Do Not Disturb | Off Steady ORANGE | DND is off DND or DND All is on |
| Do Not Disturb All | Off Flashing ORANGE Steady ORANGE | DND is off for all lines DND is on for some lines DND All is on |
| Call Forward | Off Steady ORANGE | Call forwarding is off Call forwarding is on |
| Page | Steady GREEN | Outgoing page in progress |
| Busy Lamp Field | Off Steady ORANGE Quickly flashing ORANGE Blinking ORANGE | Monitored phone is idle Monitored phone is on a call The monitored phone is ringing or has a held call* BLF registration error *Held call indication applies only to certain servers. |
| Call Handling Profile | Off Steady Orange | No CHP activated CHP activated |
| Secretarial Filtering | Off Steady ORANGE | Secretarial Filtering is off Secretarial Filtering is on |
| Hunt Group | Off Steady ORANGE | Hunt Group is off Hunt Group is on |



Network Requirements

A simple VSP861 SIP deskset installation example is shown in Figure 1. A switched network topology is recommended for your LAN.

The office LAN infrastructure should use Cat.-5/Cat.-5e cable.

The VSP861 requires a wired connection to the LAN. However, wireless connections from your LAN to other devices (such as laptops) in your office will not impede performance.

A Dynamic Host Configuration Protocol (DHCP) server is recommended and must be on the same subnet as the VSP861 desksets so that IP addresses can be auto-assigned. In most cases, your network router will have a DHCP server. By default, the VSP861 has DHCP enabled for automatic IP address assignment.



Some DHCP servers have default settings that limit the number of network IP addresses assigned to devices on the network. You should log in to your server to confirm that the IP range is sufficient.

If no DHCP server is present, you can assign a static IP to the VSP861. You can assign a static IP address using the VSP861 menu. Go to **Admin settings > Network setting > IPv4** or **IPv6 > Static** (from the list). If you do not have a DHCP server or do not manually assign static IPs, you will not be able to access the WebUI and/or enable automatic time updates from an NTP server.

A DNS server is recommended to resolve the path to the Internet and to a server for firmware and configuration updates. If necessary, the system administrator can also download upgrade files and use the WebUI to update the VSP861 firmware and/or configuration settings manually.

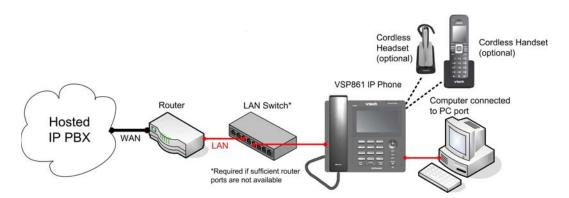


Figure 1. VSP861 Installation Example



VSP861 Configuration Methods

You can configure the VSP861 using one of the following methods:

- From the VSP861 itself, using the menus. The VSP861 menus are best suited to configuring a few settings, perhaps after the initial setup has been done. For administrators, the settings available on the VSP861 menus include network settings, account settings, and provisioning settings. See "Using the Admin Settings Menu" on page 33. Many of the settings accessible on the VSP861 are most useful for end users. Through the menu, they can customize the screen appearance, sounds, and manage calls. For more information, see the VSP861 User Guide.
- The Web User Interface, or WebUI, which you access using your Internet browser. See "Using the WebUI" on page 45. The browser-based interface is easy to navigate and best suited to configuring a large number of VSP861 settings at once. The WebUI gives you access to every setting required for configuring a single device. You can enter service provider account settings on the WebUI, configure the programmable keys, and set up provisioning, which will allow you to automatically and remotely update the VSP861 after initial configuration.
- Provisioning using configuration files. Working with configuration files is the best way to configure multiple phones. There are several methods available to enable the VSP861 to locate and upload the configuration file. For example, you can enable the VSP861, when it starts up or reboots, to check for the presence of a configuration file on a provisioning server. If the configuration file is new or has been modified in any way, the VSP861 automatically downloads the file and applies the new settings. For more information, see "Provisioning Using Configuration Files" on page 132.



Adding a Custom Logo

You can upload a custom logo to be displayed on the phone idle screen and during bootup. Uploading a logo is done using the WebUI or the configuration file. For WebUI configuration, see "Custom Image/Logo" on page 111. The configuration file parameters for uploading a custom logo are described in "Uploading a custom logo" on page 21. The default logo for bootup and idle mode is the **vtech** logo.

Logo specifications

The file type and dimensions for the logo are listed below.

File type: 24-bit colour (.bmp, .jpg, or .png)

Dimensions (w x h): Idle screenlogo: 800x480 pixels

Positioning a custom logo on the screen is a matter of creating a logo with the maximum dimensions listed above, including any surrounding white space. There are no configuration file settings to specify the x-axis or y-axis position of the logo on the screen.

Uploading a custom logo

The file.bootup_logo and file.idle_logo parameters in the configuration file allow you to upload a custom bootup logo and custom idle logo. Place the logos on your server and enter the URL for each logo for the file.idle logo and file.bootup_logo parameters.

If the downloaded logo is found to be invalid, the syslog will record one of the following errors:

- file not found
- invalid file format
- incorrect image size



Using an SD card

The micro SD card slot on the back of the VSP861 provides the following feature:

Local call recording—allows the user to record a two-way conversation during a call. The recording is stored in the available memory of the SD card.

You can configure local call recording on the desket menu, WebUI, and configuration file. See "Using the Call Recording menu" on page 40 and "Local Call Recording" on page 86.

You can view the SD card memory usage on the SD card menu (available on the deskset User settings menu) or on the WebUI status page (see "Status Page" on page 49).

Inserting and ejecting an SD card

The deskset will detect the SD card upon insertion. A splash screen indicating detection will appear for two seconds.

To eject the SD card:



- 1. When the VSP861 is idle, press **Settings** on the Home screen.
- 2. Press User settings.
- 3. On the Settings menu, scroll down and press **Storage**.
- 4. On the **SD card** menu, press **Eject**
- 5. Wait for the message It is now safe to eject your SD card, and then press down on the SD card to release it.

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CHAPTER 2

CONFIGURATION USING THE PHONE MENUS

The VSP861 Settings



application shortcut has the following menus:

- Call Settings—set DND, call forward settings, message settings, block anonymous numbers, dial anonymous, set call waiting/missed call alerts, and enable auto answer.
- App Settings—delete call history, set default screen, sort/delete directory and blacklist entries, and selecting applications to be displayed on the Home screen.
- User Settings—allows the user to set the language for the display, configure the appearance of the display, edit programmable keys, register a DECT headset and customize the audio settings.
- Admin settings—configure network settings (enter static IP addresses, for example), account settings, provisioning, security, and call recording.

This chapter contains instructions for using the Admin Settings menu and for accessing the Status menu. See the VSP861 User Guide for more information about the other menus.



Viewing the Home screen

To use the VSP861 Home screen:



- 2. Press the button for the application shortcut you want (for example, (i)).
 - Press CANCEL or to cancel an operation, exit the menu display or return to the idle screen.
 - Press to return to the Home screen.

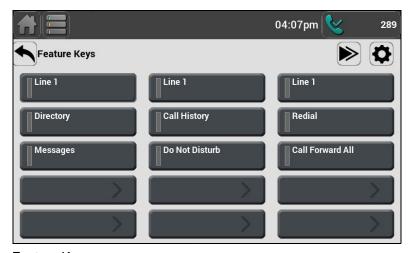


Configuring the Default screen

You can set the screen that appears when your phone is idle. Depending on your preference, you can set the screen to the Home screen (showing all the application shortcuts), the Feature keys, or the Call Manager.



Home Screen



Feature Keys



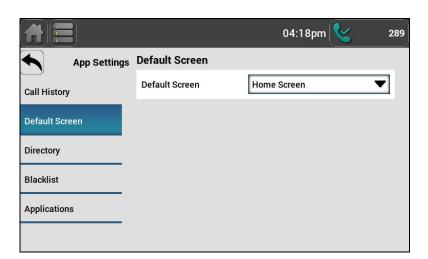


Call Manager

To configure the default screen:



- 2. Press App Settings.
- 3. From the **Settings** menu, press **Default Screen**.
- 4. Select the desired default screen from the list.





Configuring the Home screen shortcuts

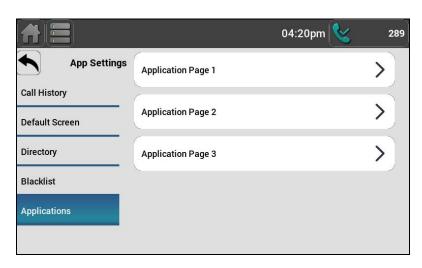
You can select and arrange the shortcuts that appear on the Home screen. The Home screen can consist of three pages, with up to 10 shortcuts per page.

To configure Home screen shortcuts:



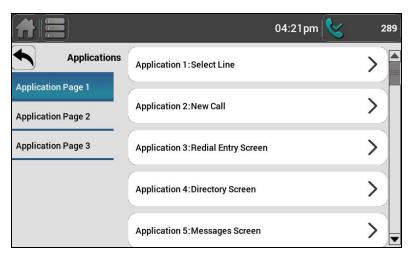
- 1. When the VSP861 is idle, press **Settings**
- 2. Press App Settings.
- 3. From the **Settings** menu, press **Applications**.

The App Settings menu appears.



4. On the App Settings menu, select the desired Home screen page.

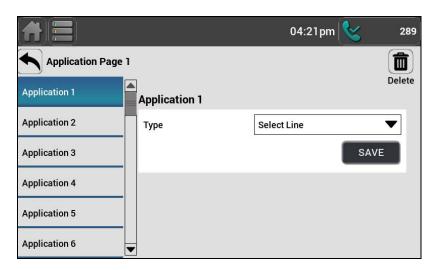
A list of 10 shortcuts for the selected page appears (numbered 1 to 10 for Application page 1, 11 to 20 for Application page 2, and 21 to 30 for Application page 3).



5. Select the numbered shortcut/application you would like to change or add (empty slots are labeled "Unassigned").



The menu for that application appears.



6. Select the desired shortcut from the list and then press **SAVE**.

Configuring the Feature keys

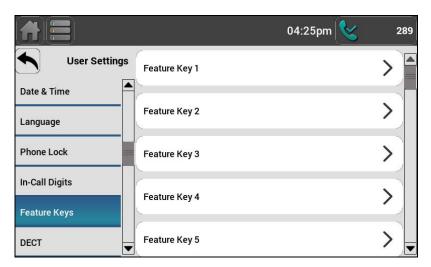
You can configure the Feature keys on the Deskset. You can also configure these keys on the WebUI (see "Programmable Feature Keys" on page 72).

To configure the Feature keys:



- Press User Settings.
- $3. \quad \text{From the User Settings menu, press } \textbf{Feature Keys}.$

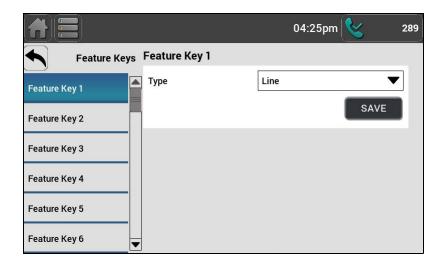
The User Settings menu appears.



4. Select the feature key to configure.



The Feature Keys menu appears.



5. Select a feature key **Type.**

Depending on the PFK type, the screen may prompt you to configure additional settings. For more information, see *"Programmable Feature Keys"* on page 72.

- 6. Enter a **Label** for the feature key.
- 7. Press Save.

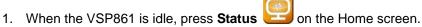


Using the Status menu

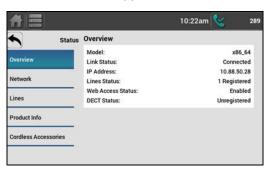
Use the **Status** menu to verify network settings and begin troubleshooting if network problems or account registration issues affect operation.

You can also find the software version of the VSP861 on the **Product Info** screen, available from the **Status** menu.

To view the Status menu:



The Status menu appears.



2. Press the desired menu.

The available status menus are listed in Table 4.

Table 4. Status menu summary

| Menu | Information listed |
|----------------|---|
| 1. Network | ■ IP type |
| (IPv4 or IPv6) | IP address |
| | Subnet Mask |
| | Prefix (IPv6 only) |
| | Gateway IP address |
| | DNS server 1 IP address |
| | DNS server 2 IP address |
| | VPN status |

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Table 4. Status menu summary

| Menu | Information listed |
|-------------------------|---|
| 2. Lines | Lines and registration status. On the Line menu, select the desired line to view detailed line status information: |
| | Line status (Registered/Not registered) |
| | Account display name |
| | Account User ID |
| | Registrar Server IP address |
| | Registrar Server port number |
| | Proxy server IP address |
| | Proxy server port number |
| 3. Product Info | Model number |
| | Serial number |
| | MAC address |
| | Boot version |
| | ■ Firmware version |
| | V-Series |
| | Hardware version |
| | ■ EMC version |
| | NTP server |
| 4. Cordless Accessories | DECT handset (Registered/Not registered) |
| | ■ DECT headset (Registered/Not registered) |



Viewing Line status

To view line status, from the **Status** menu, select **Lines**. The **Lines** menu lists the available lines, along with each line's current registration status – registered, unregistered, or disabled.

To view complete status information for a line:

• On the **Lines** menu, press the desired line. The full line status screen appears.



Using the Admin Settings Menu

To access the Admin Settings menu:

- 1. When the VSP861 is idle, press **Settings** on the Home screen.
- 2. Press Admin settings.
- 3. Use the dial pad to enter the admin password, and then press password is **admin**. The default

The Admin settings are listed in Table 5.

Table 5. Admin setting summary

| Table 6. Admin Setting Cummary | | |
|----------------------------------|------------------------------------|--|
| Setting | Options | |
| Network setting | 1. IPv4 | |
| See page 35 | 2. IPv6 | |
| | 3. VLAN | |
| Line | 1. LINE 1 | |
| See page 38 | 2. LINE 2 | |
| | 3. LINE 3 | |
| | 4. LINE 4 | |
| | 5. LINE 5 | |
| | 6. LINE 6 | |
| | 7. LINE 7 | |
| | 8. LINE 8 | |
| Provisioning | Server URL | |
| See page 39 | 2. Auth Name | |
| | 3. Auth Password | |
| Call Recording | Call Recording (Enabled, Disabled) | |
| See page 40 | 2. Record Tone (Enabled, Disabled) | |
| Phone Lock See <i>page 41</i> | Restriction Mode | |
| Remote Acccess See page 42 | Enable Web Access | |



Table 5. Admin setting summary

| Setting | Options |
|-------------------------------------|---|
| Edit password See <i>page 43</i> | 1. User PW |
| | 2. Admin PW |
| RESET | Press to display a screen that allows you to reset the phone to factory default settings. |
| RESTART | Press to display a screen that allows you to restart the phone. |

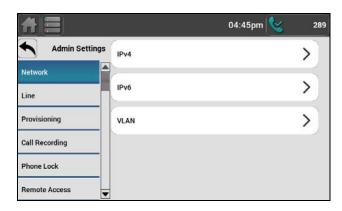


Using the Network menu

Use the Network menu to configure network-related settings for the VSP861. For more information about these settings, see "Basic Network Settings" on page 88 and "Advanced Network Settings" on page 90.

To use the Network menu:

From the Admin Settings menu, press Network.
 The Network menu appears.

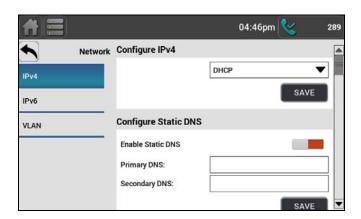


- 2. Press the desired option:
 - IPv4
 - Mode (DHCP, Static IP, PPPoE, Disable)
 - Enable Static DNS
 - Primary DNS
 - Secondary DNS
 - IPv6
 - Mode (Automatic, Static, Disable)
 - Enable Static DNS
 - Primary DNS
 - Secondary DNS
 - VLAN ID



To set the network mode:

From the Network menu, press your network type (IPv4 or IPv6).
 The selected network type screen appears (IPv4 menu shown below).



2. Select the network mode for your network type from the list, and then press Save

DHCP (IPv4) or Automatic (IPv6) is enabled by default, which means the VSP861 will get its IP address from the network. When DHCP and Automatic are disabled, you must enter a static IP address for the VSP861.



You must be familiar with TCP/IP principles and protocols to configure static IP settings.

To set static IP for the VSP861:

- 1. From the **Network** menu, press your network type (**IPv4** or **IPv6**), and then select **Static** from the list.
- 2. Enter the static **IP Address**. Use the dial pad to enter characters.
- 3. Enter the static Subnet Mask.
- 4. Enter the static Gateway.
- 5. Enter the IP address of the **Primary DNS** server.
- 6. Enter the IP address of the **Secondary DNS** server. The VSP861 uses this server if the primary server does not respond.
- 7. Press SAVE

To set static DNS:

- 1. From the Network menu, press IPv4 or IPv6.
- 2. Select Static from the list.
- 3. Touch the **Enable Static DNS** slider to turn it on



- 4. Enter the IP address for the **Primary DNS** server.
- 5. Enter the IP address for the **Secondary DNS** server.
- 6. Press **SAVE** .

To set PPPoE:

- 1. From the Network menu, press IPv4
- 2. Select **PPPoE** from the list.
- 3. Enter the Authentication Name.
- 4. Enter the Authentication Password.
- 5. Press SAVE

To set the VLAN ID for the VSP861:

- 1. From the **Network** menu, press **VLAN ID**.
- 2. .Touch the **Enable WAN VLAN** slider to enable WAN VLan.
- 3. Enter the WAN **VLAN ID**. The valid range is 0 to 4095.
- 4. Press SAVE .
- 5. Touch the **Enable PC VLAN** slider to enable PC VLan.
- 6. Enter the PC VLAN ID. The valid range is 0 to 4095.
- 7. Press SAVE .

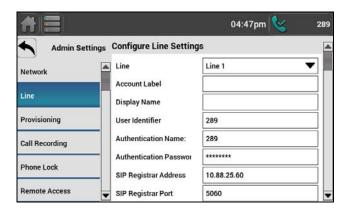


Using the Line menu

Use the **Line** menu to configure line-specific settings for the phone.

To use the Line setting menu:

From the Admin Settings menu, press Line
 The full configuration menu for that line appears.



You can configure:

- Account label
- Display name
- User Identifier
- Authentication Name
- Authentication Password
- SIP Registrar Address
- SIP Registrar port
- SIP Proxy Address
- SIP Proxy port
- Enable Line
- Answer page automatically

For more information about these settings, see "SIP Account Management" on page 51.

- 2. Edit the Line settings using the dial pad and the buttons available for each setting:
 - deletes a character
 - enables you to enter numbers, lower case letters, or upper case letters using the dial pad.
 - Save —saves and applies the new settings



Using the Provisioning menu

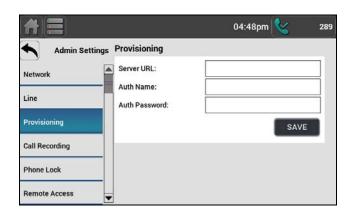
Use the Provisioning menu to configure auto-provisioning settings. For more information about auto-provisioning, see "Provisioning" on page 114 and "Provisioning Using Configuration Files" on page 132.

On the Provisioning menu you can configure:

- Server URL—the URL of the provisioning server. The URL can include a complete path to the configuration file.
- Auth Name—the username the VSP861 will use to access the provisioning server.
- Auth Password—the password the VSP861 will use to access the provisioning server.

To use the Provisioning menu:

From the Admin Settings menu, press Provisioning.
 The Provisioning menu appears.



- 2. Enter the server URL using the dial pad keys:
 - deletes a character
 - ABC —enables you to enter numbers, lower case letters, or upper case letters with the dial pad.
 - Save —prompts you to reboot the phone and apply the new settings

The format of the URL must be RFC 1738 compliant, as follows:

- 3. Enter the Auth Name for access to the provisioning server if it is not part of the server string.
- 4. Enter the Auth Password.
- 5. Press **SAVE**

[&]quot;<schema>://<user>:<password>@<host>:<port>/<url-path>"

[&]quot;<user>:<password>@" may be empty.

[&]quot;<port>" can be omitted if you do not need to specify the port number.



Using the Call Recording menu

You can configure the call recording feature on the Call Recording menu. On the Call Recording menu you can enable and disable:

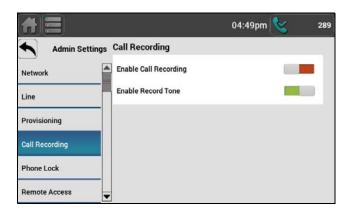
- call recording
- the call record tone (played over the line when recording begins and ends).



If you attempt to enable Call Recording with no SD card inserted, the deskset displays a **No SD card detected.** message.

To use the Call Recording menu:

From the Admin Settings menu, press Call Recording.
 The Call Recording menu appears.



- 2. Touch the **Enable Call Recording** slider to enable or disable Call Recording.
- 3. Touch the **Enable Record Tone** slider to enable or disable the Record Tone.
- 4. Press 🔨 .



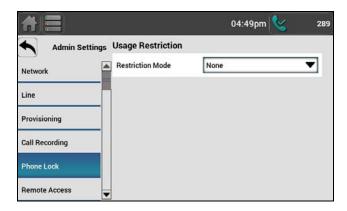
Using the Phone Lock menu

On the Phone Lock menu, you can configure the phone lock feature. The Phone Lock feature restricts certain hard keys and features unless the user enters a PIN code.

Emergency Call Only locks out all outgoing calls except emergency calls. Emergency numbers must match the Emergency Dial Plan, as configured under "General Account Settings" on page 51. Incoming calls are still allowed.

To use the Phone Lock menu:

From the Admin Settings menu, press Phone Lock.
 The Usage Restriction menu appears.



- 2. Select a **Restriction Mode** from the list.
 - None
 - Emergency Call Only
- 3. Press .

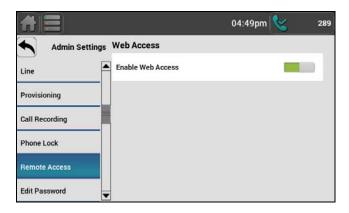


Using the Remote Access menu

On the Remote Access menu, you can enable or disable the Web server. The Web server setting determines whether the embedded WebUI is available from the phone. When the Web server is disabled, the phone can be configured using the phone menus or via provisioning only.

To use the Remote Access menu:

1. From the **Admin Settings** menu, press **Remote Access**. The **Web Access** menu appears.



- 2. Touch the **Enable Static DNS** slider to enable or disable WebUl access to the phone.
- 3. Press .



Using the Edit Password menu

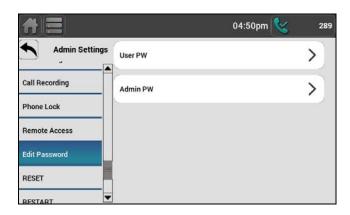
On the Edit password menu, you can reset the current User and Admin passwords.



By default, after bootup, the phone alerts you if the default passwords (user password and admin password) are still in use.

To reset the User password:

From the Admin Settings menu, press Edit Password.
 The following menu appears.



2. Press User PW.

The Edit Password screen appears.



- 3. Enter the new password using the dial pad keys.
- 4. Re-enter the new password.
- 5. Press Save

To reset the User password:

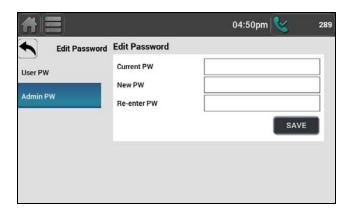
From the Admin Settings menu, press Edit Password.
 The following menu appears.





2. Press Admin PW.

The Edit Password screen appears.



- 3. Enter the current password using the dial pad keys.
- 4. Enter the new password.
- 5. Re-enter the new password.
- 6. Press Save



CHAPTER 3

USING THE WEBUI

The WebUI allows you to configure account settings, programmable keys, network settings, contact lists, and provisioning settings. The WebUI is embedded in the VSP861 operating system. When you access the WebUI, you are accessing it on the device, not on the Internet.

This chapter describes how to access the WebUI and configure VSP861 settings. This chapter covers:

- "Using the Web User Interface (WebUI)" on page 46
- "Status Page" on page 49
- "System Pages" on page 51
- "Network Pages" on page 87
- "Contacts Pages" on page 93
- "Servicing Pages" on page 106.

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Using the Web User Interface (WebUI)

The Web User Interface (WebUI) resides on the VSP861 deskset. You can access it using an Internet browser. After you log in to the WebUI, you can configure the VSP861 on the following pages:

System

- SIP Account Management (see page 51)
- Call Settings (see page 65)
- User Preferences (see *page 69*)
- Programmable Keys (see page 72)
- Speed Dial (see page 77)
- Ringer Settings (see *page 78*)
- Paging Zones (see *page 79*)
- Server Application (see *page 81*)
- Hotline Settings (see page 85)
- Local Call Recording (see *page 86*)

Network

- Basic Network Settings (see page 88)
- Advanced Network Settings (see page 90)

Contacts

- Local Directory (see page 93)
- Blacklist (see page 96)
- LDAP (see page 99)
- Broadsoft (see page 102)
- Remote XML (see *page 103*)
- Call History (see page 105)

Servicing

- Reboot (see page 106)
- Time and Date (see page 107)
- Custom Language (see page 110)
- Firmware Upgrade (see page 112)
- Provisioning (see page 114)
- Security (see page 121)
- Trusted Servers (see page 125)
- Trusted IP (see page 126)
- Certificates (see page 127)
- TR-069 (see page 129)
- System Logs (see page 130)

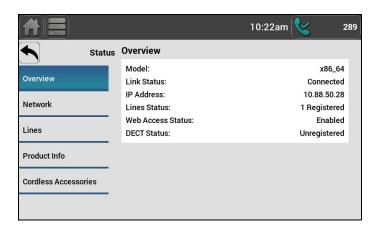
The WebUI also has a **System Status** page, where you can view network status and general information about the VSP861. The information on the Status page matches the **Status** menu available on the VSP861.

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To access the WebUI:

- Ensure that your computer is connected to the same network as the VSP861. Your computer may already be connected to the network through the PC port on the back of the VSP861.
- 2. Find the IP address of the VSP861:
 - a. When the VSP861 is idle, press **Status** on the Home screen.
 - b. On the Overview screen, note the IP Address.



- 3. On your computer, open an Internet browser. (Depending on your browser, some of the pages presented here may look different and have different controls. Ensure that you are running the latest update of your preferred browser.)
- 4. Type the VSP861 IP address in the browser address bar and press **ENTER** on your computer keyboard.

The browser displays a window asking for your user name and password.

 For the user name, enter admin. For the password, enter the default password, admin. You can change the password later on the WebUI Security page, available under Servicing.



As a security measure, the WebUI prevents you from logging in for five minutes after four (or three, depending on the browser's cache) consecutive failed log-in attempts during a five-minute period.

Click **OK**.The WebUI appears.

Click topics from the navigation bar along the top of the WebUI, and then click the links along the left to view individual pages. For your security, the WebUI times out after 10 minutes, so if it is idle for that time, you must log in again.

Most WebUI configuration pages have a save button. Click save to save changes you have made on the page. During a configuration session, click save before you move on to the next WebUI page.

The remaining procedures in this section assume that you are already logged into the WebUI.

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The settings tables in this section contain settings that appear in the WebUI, along with their equivalent settings in the configuration file template. You can use the configuration file template to create custom configuration files. Configuration files can be hosted on a provisioning server and used for automatically configuring phones. For more information, see "Provisioning Using Configuration Files" on page 132.

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Status Page

The WebUI System Status page is equivalent to the **Status** menu on the VSP861.

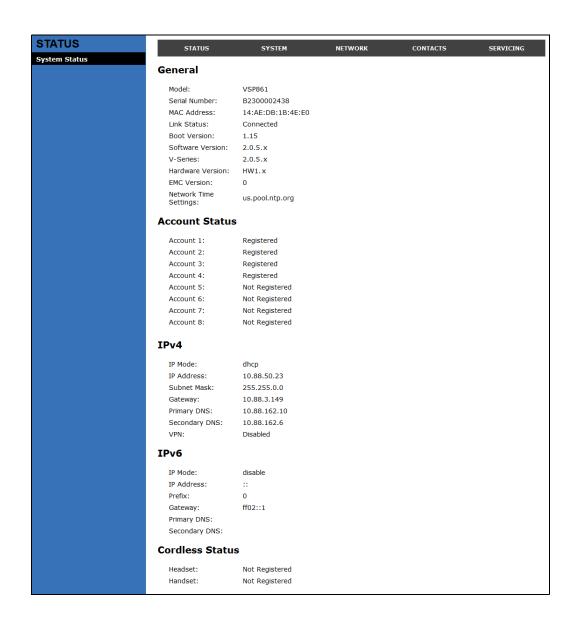
System Status

The System Status page shows:

- General information about your device, including model, MAC address, and firmware version
- Account Status information about your SIP account registration
- IPv4 and IPv6 network information regarding your device's network address and network connection
- Cordless Status indicates whether a cordless headset and/or handset is registered to the phone.

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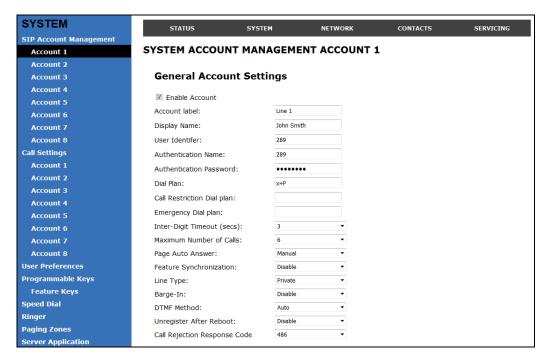


System Pages

SIP Account Management

On the SIP Account Management pages, you can configure each account you have ordered from your service provider.

The SIP Account settings are also available as parameters in the configuration file. See "sip_account Module: SIP Account Settings" on page 141.



General Account Settings

Click the link for each setting to see the matching configuration file parameter in "Configuration File Parameter Guide" on page 140. Default values and ranges are listed there.

| Setting | Description |
|----------------|---|
| Enable Account | Enable or disable the SIP account. Select to enable. |
| Account Label | Enter the name that will appear on the VSP861 display when account x is selected. |
| Display Name | Enter the Display Name. The Display Name is the text portion of the caller ID that is displayed for outgoing calls using account x. If the Account Label is blank, the Display Name appears on the VSP861 display when account x is selected. |

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| Setting | Description |
|----------------------------|---|
| User identifier | Enter the User identifier supplied by your service provider. The User ID, also known as the Account ID, is a SIP URI field used for SIP registration. Note: Do not enter the host name (e.g. "@sipservice.com"). The WebUI automatically adds the default host name. |
| Authentication name | If authentication is enabled on the server, enter the authentication name (or authentication ID) for authentication with the server. |
| Authentication password | If authentication is enabled on the server, enter the authentication password for authentication with the server. |
| Dial Plan | Enter the dial plan, with dialing strings separated by a symbol. See "Dial Plan" on page 54. |
| Call Restriction Dial Plan | Enter a call restriction dial plan, which prevents users from completing calls that match this dial plan on this account. |
| Emergency Dial Plan | Enter the Emergency Dial plan, which will apply to both restricted calls (when the phone lock feature is set to Emergency Call Only—see "Using the Phone Lock menu" on page 41 and "Security" on page 121) and other calls when the Phone Lock feature is not in use. |
| Inter Digit Timeout (secs) | Sets how long the VSP861 waits after any "P" (pause) in the dial string or in the dial plan. |
| Maximum Number of Calls | Select the maximum number of concurrent active calls allowed for that account. |
| Page Auto Answer | When set to Auto, enables the VSP861 to automatically answer when a page is received. This is usually the desired behavior for paging. |
| Feature Synchronization | Enables the VSP861 to synchronize with Broadworks Application Server. Changes to features such as DND, Call Forward All, Call Forward No Answer, and Call Forward Busy on the server side will also update the settings on the VSP861 menu and WebUI. Similarly, changes made using the VSP861 or WebUI will update the settings on the server. |
| Line Type | Select the line type to Private or Shared. A private line will be accessible only at the VSP861 you are configuring. Shared lines can be assigned to more than one VSP861. For more information about using shared lines, see the VSP861 User Guide. |
| Barge-in | Enables subscribers to shared lines to "barge in" on active calls on other shared lines. |

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| Setting | Description | |
|---------------------------------|---|--|
| DTMF method | Select the default DTMF transmission method. You may need to adjust this if call quality problems are triggering unwanted DTMF tones or you have problems sending DTMF tones in general. | |
| Unregister after reboot | Enables the phone to unregister the account(s) after rebooting-before the account(s) register again as the phone starts up. If other phones that share the same account(s) unregister unexpectedly in tandem with the rebooting VSP861, disable this setting. | |
| Call Rejection Response Code | Select the response code for call rejection. This code applies to the following call rejection cases: | |
| | User presses Reject for an incoming call (except when Call Forward Busy is enabled) | |
| | DND is enabled | |
| | Phone rejects a second incoming call with Call Waiting disabled | |
| | Phone rejects an anonymous call with Anonymous Call Rejection enabled | |
| | Phone rejects call when the maximum number of calls is reached | |

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Dial Plan

The dial plan consists of a series of dialing rules, or strings, that determine whether what the user has dialed is valid and when the VSP861 should dial the number.

There are three different types of dial plans available: general, Call Restriction, and Emergency.

In the case of overlapping dial plan definitions between Emergency, Call Restriction, and the general dial plan, the following priority (high to low) applies:

- 1. Emergency
- 2. Call Restriction
- 3. General

For example, in order to prevent a user from dialing extensions in the 9xx range while having 911 as the emergency number, use the following dial plans:

- Emergency dial plan = 911
- Call Restriction dial plan = 9xx
- General dial plan = xxx

A user who dials 920 will be unable to complete the call, and the message **920** is a restricted number will appear on the phone screen. A user who dials 911 will complete an emergency dial.



Numbers that are dialed when forwarding a call—when the user manually fowards a call, or a pre-configured number is dialed for Call Forward All, Call Forward–No Answer, or Call Forward Busy—always bypass the dial plan.

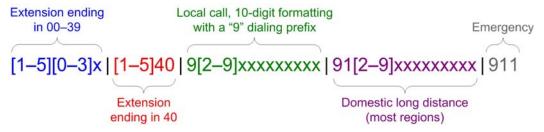
Dialing rules must consist of the elements defined in the table below.

| Element | Description |
|---------|--|
| Х | Any dial pad key from 0 to 9, including # and *. |
| [0-9] | Any two numbers separated by a hyphen, where the second number is greater than the first. All numbers within the range or valid, excluding # and *. |
| X+ | An unlimited series of digits. |
| , | This represents the playing of a secondary dial tone after the user enters the digit(s) specified or dials an external call prefix before the comma. For instance, "9,xxxxxxx" means the secondary dial tone is played after the user dials 9 until any new digit is entered. "9,3xxxxxx" means only when the digit 3 is hit would the secondary dial tone stop playing. |
| PX | This represents a pause of a defined time; X is the pause duration in seconds. For instance, "P3" would represent pause duration of 3 seconds. When "P" only is used, the pause time is the same as the Inter Digit Timeout (see "SIP Account Management" on page 51). |

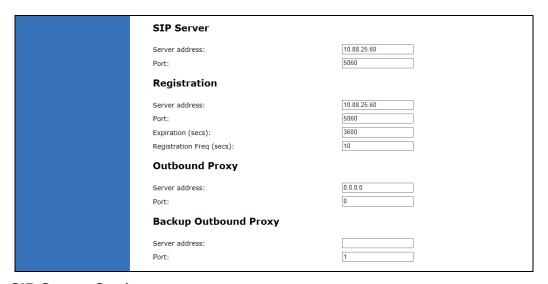


| Element | Description |
|---------|---|
| (0:9) | This is a substitution rule where the first number is replaced by the second. For example, "(4:723)xxxx" would replace "46789" with "723-6789". If the substituted number (the first number) is empty, the second number is added to the number dialed. For example, in "(:1)xxxxxxxxxxx", the digit 1 is appended to any 10-digit number dialed. |
| I | This separator is used to indicate the start of a new pattern. Can be used to add multiple dialing rules to one pattern edit box. |

A sample dial plan appears below.



See also "Prefix Dialing" on page 76.



SIP Server Settings

| Setting | Description |
|----------------|---|
| Server address | Enter the IP address or domain name for the SIP server. |
| Server port | Enter the port number that the SIP server will use. |

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Registration Settings

| Setting | Description |
|--------------------------|--|
| Server address | Enter the IP address or domain name for the registrar server. |
| Server port | Enter the port number that the registrar server will use. |
| Expiration | Enter the desired registration expiry time in seconds. |
| Registration Freq (secs) | Enter the desired registration retry frequency in seconds. If registration using the Primary Outbound Proxy fails, the Registration Freq setting determines the number of seconds before a registration attempt is made using the Backup Outbound Proxy. |

Outbound Proxy Settings

| Setting | Description |
|----------------|---|
| Server address | Enter the IP address or domain name for the proxy server. |
| Server port | Enter the port number that the proxy server will use. |

Backup Outbound Proxy Settings

| Setting | Description |
|----------------|--|
| Server address | Enter the IP address or domain name for the backup proxy server. |
| Server port | Enter the port number that the backup proxy server will use. |

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| Caller Identity | |
|------------------------------------|----------|
| Source Priority 1: | PAI • |
| Source Priority 2: | RPID • |
| Source Priority 3: | From |
| Audio | |
| Addio | |
| Codec Priority 1: | G.711u • |
| Codec Priority 2: | G.711a |
| Codec Priority 3: | G.729a/b |
| Codec Priority 4: | G.726 |
| Codec Priority 5: | G.722 |
| Codec priority 6: | None - |
| Codec priority 7: | iLBC |
| ■ Enable Voice Encryption (SRTP) | |
| Enable G.729 Annex B | |
| Preferred Packetization Time (ms): | 20 • |
| DTMF Payload Type: | 101 |
| Quality of Service | |
| DSCP (voice): | 46 |
| | 26 |
| Signaling Settings | |
| Local SIP Port: | 5060 |
| Transport: | UDP - |
| Voice | |
| Min Local RTP Port: | 18000 |
| | 19000 |

Caller Identity Settings

| Setting | Description |
|-------------------|--|
| Source Priority 1 | Select the desired caller ID source to be displayed on the incoming call screen: "From" field, RPID (Remote-Party ID) or PAI (P-Asserted Identity) header. |
| Source Priority 2 | Select the lower-priority caller ID source. |
| Source Priority 3 | Select the lowest-priority caller ID source. |

Audio Settings

| Setting | Description |
|------------------|---|
| Codec priority 1 | Select the codec to be used first during a call. |
| Codec priority 2 | Select the codec to be used second during a call if the previous codec fails. |
| Codec priority 3 | Select the codec to be used third during a call if the previous codec fails. |
| Codec priority 4 | Select the codec to be used fourth during a call if the previous codec fails. |

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| Setting | Description |
|--------------------------------------|---|
| Codec priority 5 | Select the codec to be used fifth during a call if the previous codec fails. |
| Codec priority 6 | Select the codec to be used sixth during a call if the previous codec fails. |
| Codec priority 7 | Select the codec to be used last during a call if the previous codec fails. |
| Enable voice encryption (SRTP) | Select to enable secure RTP for voice packets. |
| Enable G.729 Annex B | When G.729a/b is enabled, select to enable G.729 Annex B, with voice activity detection (VAD) and bandwidth-conserving silence suppression. |
| Preferred Packetization Time (ms) | Select the packetization interval time. |
| DTMF Payload Type | Set the DTMF payload type for in-call DTMF from 96–127. |

Quality of Service

| Setting | Description |
|------------------|---|
| DSCP (voice) | Enter the Differentiated Services Code Point (DSCP) value from the Quality of Service setting on your router or switch. |
| DSCP (signaling) | Enter the Differentiated Services Code Point (DSCP) value from the Quality of Service setting on your router or switch. |

Signaling Settings

| Setting | Description |
|----------------|--|
| Local SIP port | Enter the local SIP port. |
| Transport | Select the SIP transport protocol: |
| | UDP (User Datagram Protocol) is generally less prone to latency, but SIP data may be subject to network congestion. |
| | TCP (Transmission Control Protocol) is the most reliable protocol and includes error checking and delivery validation. |
| | ■ TLS (Transport Layer Security)—the VSP861 supports secured SIP signaling via TLS. Optional server authentication is supported via user-uploaded certificates. TLS certificates are uploaded using the configuration file. See "file Module: Imported File Parameters" on page 201 and consult your service provider. |

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Voice

| Setting | Description |
|--------------------|--|
| Min Local RTP port | Enter the lower limit of the Real-time Transport Protocol (RTP) port range. RTP ports specify the minimum and maximum port values that the phone will use for RTP packets. |
| Max Local RTP port | Enter the upper limit of the RTP port range. |

| Paging: |
|-----------------------------|
| |
| Call Park: |
| Parked Call Retrieval: |
| Voicemail: |
| DND ON: |
| DND OFF: |
| Call Forward All ON: |
| Call Forward All OFF: |
| Call Forward No Answer ON: |
| Call Forward No Answer OFF: |
| Call Forward Busy ON: |
| Call Forward Busy OFF: |
| Anonymous Call Reject ON: |
| Anonymous Call Reject OFF: |
| Anonymous Call ON: |
| Anonymous Call OFF: |
| Call Waiting ON: |
| Call Waiting OFF: |
| Group Call Pickup: |
| Direct Call Pick Up: |
| Hunt Group Sign ON: |
| Hunt Group Sign OFF: |
| Secretarial Filtering ON: |
| Secretarial Filtering OFF: |

Feature Access Codes Settings

If your IP PBX service provider uses feature access codes, then enter the applicable codes here. You can assign many of these features to programmable feature keys, which enables end users to press the keys to dial out the codes you enter here. To configure programmable feature keys, see "Programmable Feature Keys" on page 72.

| Setting | Description | Assignable to PFK? |
|-----------|--|--------------------|
| Paging | Enter the paging access code. | Yes |
| Call Park | Enter the call park access code. Broadsoft provides a feature access code for the park feature. Asterisk/Metaswitch provides a parking lot extension number for the park feature. Enter the parking lot extension number here. | Yes |

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| Setting | Description | Assignable to PFK? |
|-------------------------------|--|--------------------|
| Parked Call Retrieval | Enter the call park retrieval access code. Broadsoft and Asterisk/Metaswitch provide a feature access code for park retrieval. | Yes |
| Voicemail | Enter the voicemail access code. The code is dialed when the user selects a line from the phone's Message menu. | Yes |
| DND ON | Enter the Do Not Disturb ON access code. | Yes |
| DND OFF | Enter the Do Not Disturb OFF access code. | Yes |
| Call Forward All ON | Enter the Call Forward All ON access code. As a Comverse feature, the code supports value substitution. For example, if an ON code is configured as *71%N#, %N will be replaced by another value before it is used as the INVITE target. | Yes |
| Call Forward All OFF | Enter the Call Forward All OFF access code. As a Comverse feature, the code supports value substitution. | Yes |
| Call Forward No Answer ON | Enter the Call Forward No Answer ON access code. | Yes |
| Call Forward No Answer OFF | Enter the Call Forward No Answer OFF access code. | Yes |
| Call Forward Busy ON | Enter the Call Forward Busy ON access code. | Yes |
| Call Forward Busy OFF | Enter the Call Forward Busy OFF access code. | Yes |
| Anonymous Call Reject ON | Enter the Anonymous Call Reject ON access code. | No |
| Anonymous Call Reject OFF | Enter the Anonymous Call Reject OFF access code. | No |
| Anonymous Call ON | Enter the Anonymous Call ON access code. | No |
| Anonymous Call OFF | Enter the Anonymous Call OFF access code. | No |
| Call Waiting ON | Enter the Call Waiting ON access code. | No |
| Call Waiting OFF | Enter the Call Waiting OFF access code. | No |
| Group Call Pickup | Enter the Group Call Pickup code. Dialing the code enables the user to answer a call ringing at another VSP861 that is part of the same group. | Yes |
| Direct Call Pickup | Enter the Direct Call Pickup code. Dialing the code enables the user to answer a call ringing at another VSP861. | Yes |



| Setting | Description | Assignable to PFK? |
|---------------------------|---|--------------------|
| Hunt Group Sign ON | Enter the Comverse Hunt Group ON code. Supports value substitution. | Yes |
| Hunt Group Sign OFF | Enter the Comverse Hunt Group OFF code. Supports value substitution. | Yes |
| Secretarial Filtering ON | Enter the Comverse Secretarial Filtering ON code. Supports value substitution. | Yes |
| Secretarial Filtering OFF | Enter the Comverse Secretarial Filtering OFF code. Supports value substitution. | Yes |

| Busy Lamp Field | |
|------------------------------|------|
| List URI: | |
| Remote Pickup Code: | |
| BLF Subscription Expiration: | 3600 |
| Voicemail Settings | |
| Enable MWI Subscription | |
| Mailbox ID: | |
| Expiration (secs): | 3600 |
| ☐ Ignore Unsolicited MWI | |
| ■ Enable Stutter Dial Tone | |
| NAT Traversal | |
| ☐ Enable STUN | |
| Server Address: | |
| Port: | 3478 |
| ■ Enable STUN Keep-Alive | |
| Keep-Alive Interval (sec): | 30 |

Busy Lamp Field

| Setting | Description |
|-----------------------------|---|
| List URI | Enter the BLF list URI, as supplied by or set up with your service provider. For example, blf-list1@sipservice.com. This list contains a list of extensions that are eligible for BLF monitoring. You can assign keys for BLF monitoring on the Programmable Keys page. See "Programmable Feature Keys" on page 72. |
| Remote Pickup Code | Enter the remote pickup code for the BLF list, as supplied by your service provider. |
| BLF subscription expiration | Enter the BLF subscription expiry time (in seconds) for account x. |

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Voicemail Settings

| Setting | Description | |
|-----------------------------|--|--|
| Enable MWI Subscription | When enabled, the account subscribes to the "message summary" event package. The account may use the User ID or the service provider's "Mailbox ID". | |
| Mailbox ID | Enter the URI for the mailbox ID. The phone uses this URI for the MWI subscription. If left blank, the User ID is used for the MWI subscription. | |
| MWI subscription expiration | Enter the MWI subscription expiry time (in seconds) for account x. | |
| Ignore unsolicited MWI | When selected, unsolicited MWI notifications—notifications in addition to, or instead of SUBSCRIBE and NOTIFY methods—are ignored for account x. If the VSP861 receives unsolicited MWI notifications, the Message Waiting LED will not light to indicate new messages. Disable this setting if: | |
| | MWI service does not involve a subscription to a voicemail server. That is, the server supports unsolicited MWI notifications. | |
| | you want the Message Waiting LED to indicate new messages when the VSP861 receives unsolicited MWI notifications. | |
| Enable Stutter Dial Tone | Enables or disables the stutter dial tone for that account (indicating message(s) waiting) when the phone goes off hook. | |

NAT Traversal

| Setting | Description |
|----------------------------|--|
| Enable STUN | Enables or disables STUN (Simple Traversal of UDP through NATs) for account x. The Enable STUN setting allows the VSP861 to identify its publicly addressable information behind a NAT via communicating with a STUN server. |
| Server address | Enter the STUN server IP address or domain name. |
| Server port | Enter the STUN server port. |
| Enable STUN Keep-Alive | Enables or disables STUN keep-alives. Keep-alive packets are used to maintain connections established through NAT. |
| Keep-alive interval (secs) | Enter the interval (in seconds) for sending keep-alives. |

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| | | _ |
|---|------|---|
| Music On Hold | | |
| ☑ Enable Local MoH | | |
| Network Conference | • | |
| ☐ Enable Network Conference Conference URI: | | |
| Session Timer | | |
| ☐ Enable Session Timer | | |
| Minimum Value (secs): | 90 | |
| Maximum Value (secs): | 1800 | |
| Jitter Buffer | | |
| Fixed | | |
| Fixed Delay (ms): | 80 | |
| Adaptive | | |
| Normal Delay (ms): | 80 | |
| Minimum Delay (ms): | 60 | |
| Maximum Delay (ms): | 240 | |
| Keep Alive | | |
| ☐ Enable Keep Alive | | |
| Keep Alive interval (secs): | 15 | |
| Ignore Keep Alive Failure | | |
| Save | | |
| | | |

Music on Hold Settings

| Setting | Description |
|------------------|---|
| Enable Local MoH | Enables or disables a hold-reminder tone that the user hears when a far-end caller puts the call on hold. |

Network Conference Settings

| Setting | Description |
|------------------------------|--|
| Enable Network Conference | Enables or disables network conferencing for account x. |
| Conference URI | Enter the URI for the network bridge for conference handling on account x. |

Session Timer

| Setting | Description |
|----------------------|---|
| Enable Session Timer | Enables or disables the SIP session timer. The session timer allows a periodic refreshing of a SIP session using the RE-INVITE message. |
| Minimum value (secs) | Sets the session timer minimum value (in seconds) for account x. |
| Maximum value (secs) | Sets the session timer maximum value (in seconds) for account x. |

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Jitter Buffer

| Setting | Description |
|--------------------|--|
| Fixed | Enable fixed jitter buffer mode. |
| Fixed Delay (ms) | If Fixed is selected, enter the fixed jitter delay. |
| Adaptive | Enable adaptive jitter buffer mode. |
| Normal Delay (ms) | If Adaptive is selected, enter the normal or "target" delay. |
| Minimum Delay (ms) | Enter the minimum delay. |
| Maximum Delay (ms) | Enter the maximum delay. This time, in milliseconds, must be at least twice the minimum delay. |

Keep Alive

| Setting | Description |
|------------------------------|--|
| Enable Keep Alive | Enable SIP keep alive in service of NAT traversal and as a heartbeat mechanism to audit the SIP server health status. Once enabled, OPTIONS traffic should be sent whenever the account is registered. OPTIONS traffic will occur periodically according to the keep-alive interval. |
| Keep Alive Interval (secs) | Set the interval at which the OPTIONS for the keep-alive mechanism are sent. |
| Ignore Keep Alive Failure | Enable the phone to ignore keep-alive failure, if the failure can trigger account re-registration and re-subscription (and active calls are dropped). |

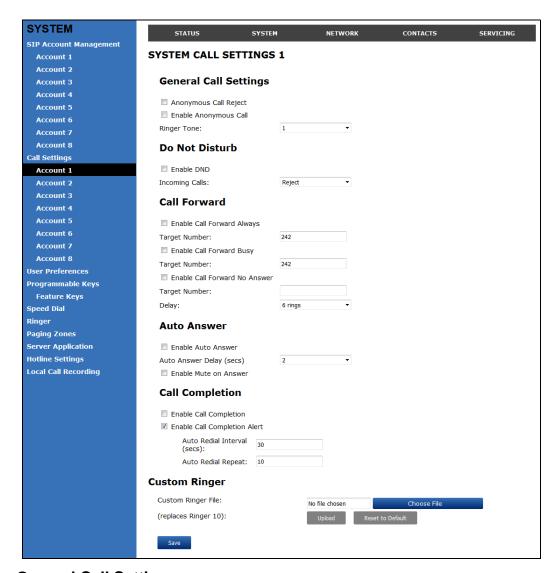
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Call Settings

You can configure call settings for each account. Call Settings include Do Not Disturb and Call Forward settings.

The call settings are also available as parameters in the configuration file. See "call_settings Module: Call Settings" on page 188.



General Call Settings

| Setting | Description |
|-----------------------|--|
| Anonymous Call Reject | Enables or disables rejecting calls indicated as "Anonymous." |
| Enable Anonymous Call | Enables or disables outgoing anonymous calls. When enabled, the caller name and number are indicated as "Anonymous." |

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| Setting | Description |
|-------------|---|
| Ringer Tone | Sets the ringer tone for incoming calls on the account. |

Do Not Disturb

| Setting | Description |
|----------------|--|
| Enable DND | Turns Do Not Disturb on or off. |
| Incoming calls | When set to Show, the phone displays incoming call information while Do Not Disturb is on. When set to Reject, the phone rejects incoming calls without alerting the user. |

Auto Answer

| Setting | Description |
|-----------------------|---|
| Enable Auto Answer | Enables or disables unconditional Auto Answer. Auto Answer allows a deskset or conference phone to automatically answer incoming calls to that account without user intervention. An auto answer tone will sound. |
| Auto Answer Delay | Sets the delay before the phone auto answers a call. Before the phone auto answers, the incoming call behaves identical to a normal call. Unless the user responds to the call (with reject, forward, answer, etc.), the phone answers the call after the delay expires. If Auto Answer Delay is set to zero, the incoming call is answered right away without triggering a ringer tone or ringer splash. However, the auto answer tone is still audible. |
| Enable Mute on Answer | Enables or disables muting the mic upon auto answering. Enabling muting is useful if the auto answered call is for the purpose of a one-way announcement. The user can unmute the call any time after being auto answered. |
| Ringer Tone | Sets the ringer tone for incoming calls on the account. |

Call Forward

| Setting | Description |
|-------------------------------|---|
| Enable Call Forward Always | Enables or disables call forwarding for all calls on that line. Select to enable. |
| Target Number | Enter a number to which all calls will be forwarded. |

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| Setting | Description |
|----------------------------------|--|
| Enable Call Forward Busy | Enables or disables forwarding incoming calls to the target number if: |
| | the number of active calls has reached the maximum number of calls configured for account x |
| | Call Waiting Off is selected. |
| Target Number | Enter a number to which calls will be forwarded when Call Forward Busy is enabled. |
| Enable Call Forward No Answer | Enables or disables call forwarding for unanswered calls on that line. |
| Target Number | Enter a number to which unanswered calls will be forwarded. |
| Delay | Select the number of rings before unanswered calls are forwarded. |

Call Completion

When the user calls a busy number, the Call Completion feature enables the phone to redial the busy number automatically. You can configure the redialing to take place after a set interval and for a set number of times.

Call completion settings must be configured for each account.

| Setting | Description |
|---------------------------------|--|
| Enable Call Completion | Enable or disable the call completion feature. Calls to busy numbers will prompt a "Retry later?" message on the phone screen. |
| Enable Call Completion Alert | Enables or disables an audible alert (similar to a hold reminder alert tone) if the user is on another call when the auto redial interval expires. |
| Auto Redial Interval (secs) | Sets the countdown timer until the user is prompted for the next dialing attempt. |
| Auto Redial Repeat | Sets how many auto redial attempts are made. |

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Custom Ringer

| Setting | Description |
|--------------------|--|
| Custom Ringer File | Upload a custom ringer audio file. This ringer replaces the factory default ringer 10. The maximum file size is 300 k. The following WAV format is accepted: |
| | Audio sample rate: 16 kHz |
| | Audio sample size: 16 bit |
| | Channels: 1 (mono) |
| | Audio format: PCM, Signed 16bit, Little Endian |

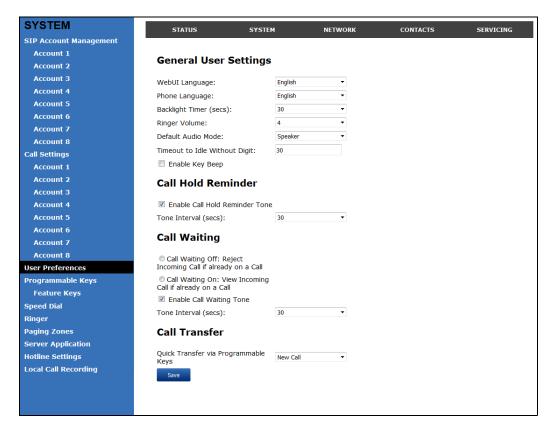
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User Preferences

On the User Preferences page, you can configure some basic settings for the phone and set hold reminder and call waiting settings. The User Preferences page is also available to phone users when they log on to the WebUI.

The user preference settings are also available as parameters in the configuration file. See "user_pref Module: User Preference Settings" on page 184.



General User Settings

Click the link for each setting to see the matching configuration file parameter in "Configuration File Parameter Guide" on page 140. Default values and ranges are listed there.

| Setting | Description |
|------------------------|---|
| WebUI Language | Sets the language that appears on the WebUI. |
| Phone Language | Sets the language that appears on the phone. Other languages may be added in a future release. |
| Backlight Timer (secs) | Sets how long (in seconds) the screen backlight stays on after the last button press. |
| Ringer Volume | Sets the ringer volume for incoming calls. You can also use the VOLUME ▼ or ▲ keys on the VSP861. |

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| Setting | Description |
|-------------------------------|---|
| Default Audio Mode | Sets how calls are answered when users press a line key or Accept. |
| Timeout to Idle Without Digit | Sets the timeout (in seconds) after the phone goes off hook and no digits are entered. After the timeout, the phone returns to idle mode. |
| Enable Key Beep | Enables or disables key-press beeps. |

Call Hold Reminder

| Setting | Description |
|-----------------------------------|--|
| Enable Call Hold Reminder Tone | Enables or disables the call hold reminder tone. Select to enable. |
| Tone Interval (secs) | Sets the interval for the call hold reminder tone, in seconds. |

Call Waiting

| Setting | Description |
|--------------------------|--|
| Call Waiting Off | When selected, disables incoming call notifications when the user is already on a call. Incoming calls are rejected. Incoming callers hear a busy signal. When Call Waiting Off is selected, and Call Forward Busy is enabled, incoming calls are handled according to the Call Foward Busy setting. |
| Call Waiting On | When selected, enables incoming call notifications when the user is already on a call. |
| Enable Call Waiting Tone | Enables or disables the call waiting tone. Select to enable. |
| Tone Interval (secs) | Sets the interval for the call waiting tone, in seconds. |

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Call Transfer

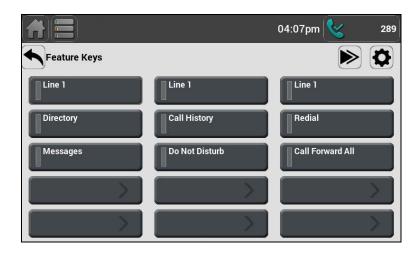
| Setting | Description |
|---|--|
| Quick Transfer via Programmable Keys | Sets transfer options for Quick Dial and BLF Programmable keys during an active call. When a quick dial key or BLF key is pressed during an active call, the key will: Start a new call |
| | Perform a blind transfer of the active call to the extension associated with the quick dial or BLF key. This enables one-button operation for Call Park, Park Retrieval, and Park Orbit monitoring. |
| | Perform an attended transfer of the active call to the extension associated with the quick dial or BLF key. |
| | The above operations are server dependent. The server has to support all of the following with Call Park to make one-button operation possible: |
| | Monitoring a park orbit as an extension via BLF subscription (rfc4235) |
| | Remote BLF pickup via one of the two pickup options: |
| | New call via SIP INVITE |
| | Dialog based via SIP INVITE with REPLACE |
| | Parking an active call via blind transfer to a park orbit. |

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Programmable Feature Keys

The Programmable Keys pages allow you to program the Feature Keys Page on the deskset.



You can assign up to 30 functions to the keys listed on the Feature Keys page.

Keys can have identical functions, depending on the "Type" of key. For example, you can assign several **Line** keys to Account 1 to enable users to manage multiple calls on Account 1. You can also assign multiple **Quick Dial** keys.

The programmable key settings are also available as parameters in the configuration file. See "pfk Module: Programmable Feature Key Settings" on page 192.

For the programmable key default settings, see "Programmable Feature Keys" on page 15.

Click the link for each setting to see the matching configuration file parameter in "Configuration File Parameter Guide" on page 140. Default values and ranges are listed there.

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| SYSTEM | STATUS | SYSTEM | NETWORK | CONTACTS | SERVICING |
|------------------------|------------|------------------|----------|----------|-----------|
| SIP Account Management | | | | | |
| Account 1 | Programmab | le Keys | | | |
| Account 2 | Key | Туре | Label | Value | Account |
| Account 3 | Key 1 | Line | ▼ | | Account 1 |
| Account 4 | Key 2 | Line | ~ | | Account 1 |
| Account 5 | | | | | |
| Account 6 | Key 3 | Line | • | | Account 2 |
| Account 7 | Key 4 | Directory | <u> </u> | | Account 1 |
| Account 8 | Key 5 | Call History | • | | Account 1 |
| Call Settings | Key 6 | Redial | • | | Account 1 |
| Account 1 | Key 7 | Messages | • | | Account 1 |
| Account 2 | Key 8 | Do Not Disturb | • | | Account 1 |
| Account 3 | Key 9 | Call Forward All | - | | Account 1 |
| Account 4 | Key 10 | N/A | - | | Account 1 |
| Account 5 | · · | | | | |
| Account 6 | Key 11 | N/A | <u> </u> | | Account 2 |
| Account 7 | Key 12 | N/A | • | | Account 1 |
| Account 8 | Key 13 | N/A | • | | Account 1 |
| User Preferences | Key 14 | N/A | • | | Account 1 |
| Programmable Keys | Key 15 | N/A | - | | Account 1 |
| Feature Keys | Key 16 | N/A | • | | Account 1 |
| Speed Dial | Key 17 | N/A | • | | Account 1 |
| Ringer Danian Zanan | · · | | • | | |
| Paging Zones | Key 18 | N/A | | | Account 1 |
| Server Application | Key 19 | N/A | - | | Account 1 |
| Hotline Settings | Key 20 | N/A | • | | Account 1 |
| Local Call Recording | Key 21 | N/A | • | | Account 1 |

| Programmable Key Type | Description |
|-----------------------|--|
| Line | Configures the key for accessing an account. Users can make or answer calls by pressing these keys. The key LED will change according to call activity. After selecting Account in the Type column, select the account number in the Account column. |
| Directory | Configures the key to access the Directory menu. Users can then press the key to view the Directory menu. |
| Call History | Configures the key to access the Call History list. Users can then press the key to view the Call History list. |
| Redial | Configures the key to access the Redial list. Users can then press the key to view the Redial list. |
| Messages | Configures the key to access the Message menu. Users can then press the key to view the Message menu. |
| Do Not Disturb | Configures the key to turn Do Not Disturb on or off for a selected account. The key is lit orange when DND is on. If DND is on and a Do Not Disturb All key is also available, the DND All key flashes to indicate that not all accounts have DND set. |

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| Programmable Key Type | Description |
|------------------------|--|
| Do Not Disturb All | Configures the key to turn Do Not Disturb on or off for all accounts. The key is lit orange when DND All is on (as are any other DND keys). If one or more accounts also has a dedicated DND key, turning DND off for a particular account will cause the DND All key to flash, indicating that not all accounts have DND set. |
| Call Forward All | Configures the key to turn Call Forward All on or off. In the Account column, select the account for which Call Forward All will apply. Before assigning the key, ensure that you configure Call Forward settings on the Call Settings page. |
| Call Forward No Answer | Configures the key to turn Call Forward No Answer on or off. In the Account column, select the account for which Call Forward No Answer will apply. Before assigning the key, ensure that you configure Call Forward settings on the Call Settings page. |
| Call Forward Busy | Configures the key to turn Call Forward Busy on or off. In the Account column, select the account for which Call Forward Busy will apply. Before assigning the key, ensure that you configure Call Forward settings on the Call Settings page. |
| Quick Dial | Configures the key to dial a number on the selected line. After selecting Quick Dial, enter the number to be dialed in the Value column. In the Account column, select the account on which the number will be dialed out. |
| BLF (Busy Lamp Field) | Configures the key to monitor another extension. In the Value column, enter the URI of the extension you want to monitor with this key. For example, 2325552001@sipservice.com. For configuring BLF interoperability when using certain service platforms, see sip_account.x.blf_variant. |
| XML App | Configures the key to open the XML browser. Pressing the key initiates an HTTP(s) GET request to the server. Enter the URI of the XML application to be executed. |
| Page | Configures the key to call one or a group of phones. Pressing the key dials the Paging feature access code. You must enter the feature access code for Paging on the SIP Account Management page. For some service providers, you must also enter a page extension value in the Value column. This value will be dialed along with the Paging feature access code. You can configure pages to be automatically answered. See "Page Auto Answer" under "SIP Account Management" on page 51. |

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| Programmable Key Type | Description |
|-----------------------|---|
| Multicast Page | Configures the key to make outgoing multicast pages. In the Value column, enter a valid Paging Zone ID (ranging from 1 to 10). Multicast paging differs from standard paging in that it is handled locally by the VSP861 and does not require a subscription through the hosted server. To use multicast paging, you must first set up paging zones on the WebUI. See "Paging Zones" on page 79. See also pfk.x.multicast_zone in the configuration file. |
| Park Call | Enables the user to park a call. Pressing the key dials the Call Park feature access code (FAC). You must enter the feature access code for Call Park on the SIP Account Management page. For some service providers, you must also enter a park extension in the Value column. This value will be dialed along with the Call Park FAC. See also sip_account.x.park_variant. |
| Retrieve Parked Call | Enables the user to retrieve a parked call. Pressing the key dials the Parked Call Retrieval feature access code (FAC) configured on the SIP Account Management page. For some service providers, you must also enter a park retrieval extension in the Value column. This value will be dialed along with the Parked Call Retrieval FAC. |
| In Call DTMF | Configures the key to dial a string of numbers while the end user is on a call. For example, pressing the key might dial a conference access code. After selecting In Call DTMF, enter the number to be dialed in the Value column. See also pfk.x.incall_dtmf in the configuration file. |
| Call Return | Configures the key to dial the number of the last missed call. |
| Group Call Pickup | Enables the user to answer a call ringing at another extension. The call can be ringing at any extension in the phone's call pickup group. Pressing the key dials the Group Call Pickup feature access code (FAC) configured on the SIP Account Management page. |
| Direct Call Pickup | Enables the user to answer a call ringing at a specific VSP861 or compatible SIP telephone. Pressing the key dials the Direct Call Pickup feature access code (FAC) configured on the SIP Account Management page. Depending on the server requirements, the user may then need to enter the number of the ringing extension. |

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| Programmable Key Type | Description |
|-----------------------|---|
| Prefix Dialing | Configures the key for prefix dialing. Pressing a PFK assigned to Prefix Dialing will automatically go off-hook into Dial mode using the account configured for the PFK. The digits entered under Value are concatenated to any number that the user enters. The outgoing call will include the [Prefix] + [Dialing string]. The Prefix digits are hidden from the user. The prefix digits are visible during Dialing and Call Active states. Note that %N can be used for substitution of user-entered digits. For example, * 71%N# will use [*71] + [user-entered digits] + [#] as the outgoing dialing string. The dial plan is enabled after the user enters one or more digits. Both prefix digits (hidden) and user-entered digits (visible) are used for dial plan matching. The timeout-to-dial element in the dial plan is suspended if the user navigates away from the dialing screen. |
| Flash | Configures the key as a Flash key. With one call active, pressing Flash puts the active call on hold and displays a new call screen with live dial. With two calls active, pressing Flash: |
| | puts the active call on hold and retrieves a held call, or |
| | answers an incoming call. |
| Call Handling Profile | Configures the key for Comverse call handling profile. Enter the string of the call handling profile that the PFK LED will indicate. |
| Hunt Group | Configures the key to turn Comverse hunt group on or off. Enter the hunt group extension number assigned for this key. |
| Secretarial Filtering | Configures the key to turn Comverse secretarial filtering on or off. Enter the manager's extension number assigned for this key. |
| Phone Lock | Configures the key to enable or disable the phone lock. For more information, see "Using the Phone Lock menu" on page 41 and "Security" on page 121. |

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Speed Dial Keys

On the Speed Dial page, you can enter up to 10 speed dial numbers. For each speed dial number you enter, you must assign the account on which the number will be dialed out.

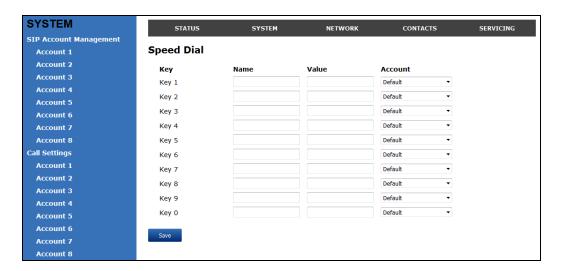
To dial a speed dial number, press and hold the dial pad key that matches the speed dial entry number.



This menu duplicates the speed dial menu on the phone (**Main Menu > Features > Speed dial**). Entries that are entered and saved on the WebUI replace entries that were entered using the phone. Similarly, entries that are configured using the phone menu will update entries on the WebUI.

The speed dial key settings are also available as parameters in the configuration file. See "speed_dial Module: Speed Dial Settings" on page 196.

After entering information on this page, click to save it.



To enter speed dial numbers:

- 1. In the **Name** column, enter the name associated with this speed-dial entry.
- 2. In the Value column, enter a phone number for the desired key.
- 3. In the **Account** column, select the account that this speed dial number will use.
- 4. Click Save .

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Speed Dial Keys

Click the link for each setting to see the matching configuration file parameter in "speed_dial Module: Speed Dial Settings" on page 196.

| Setting | Description |
|---------|---|
| Name | The name associated with the speed dial entry. |
| Value | The phone number that the speed dial key dials when pressed and held. |
| Account | The SIP account that the phone will use to dial the number. |

Ringer Settings

The Ringer Settings enable you to provide a distinctive ringing feature via the custom Alert-Info header associated with an incoming call. This setting overrides the ringer tone you have set for the account. For example, you can set a unique ringer tone to alert the VSP861 user upon receiving any incoming calls tagged as "important" in the Alert-Info header.

The SIP Invite message contains an Alert-Info header that the phone checks in order to determine which ringer tone to play. The Alert-Info header format is as follows:

Alert-Info: info=info_text

If the header contains the "info" parameter, the phone attempts to match it to the Distinctive Ringing Text. If there is a match, the associated tone will play. If there is no match, the default tone for the account will play.

The matching is done on a "first match" basis. In the case of duplicate text strings, the ringer tone associated with the first matched entry in the Distinctive Ringing Text list will play.

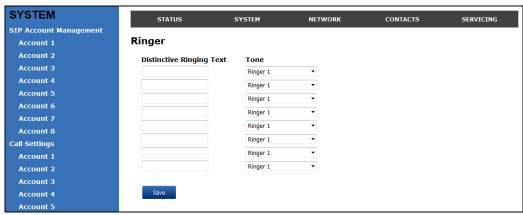
The server-side configuration must be done with your service provider. The SIP Invite text ("Distinctive Ringing Text" on the Ringer WebUI page) must be entered in the format **ringerx**, where x is the ringer number from 1 to 10. For example, to match Ringer 1 enter **ringer1**.

The ringer settings are also available as parameters in the configuration file. See *"ringersetting Module: Distinctive Ringer Settings" on page 199.*

After entering information on this page, click Save to save it.

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| Setting | Description |
|--------------------------|---|
| Distinctive Ringing Text | Enter the text that will match the "info" parameter and play the ringer tone. The Distinctive Ringing Text must be in the format ringerx where x is the ringer tone 1 to 10. |
| Tone | Select the desired ringer from the list. |

Paging Zones

On the Paging Zones page, you can enter the multicast IP addresses that the phone will monitor. When a page is sent out using this multicast IP address, all phones that are programmed to monitor that IP address will receive the paging RTP stream and play the page on their speakerphone. You can also enable the phone to send out multicast pages using a particular multicast IP address.

You must first set up paging groups (each group consisting of a multicast paging IP address and assigned User IDs) on your SIP PBX. The VSP861 can monitor a maximum of 10 multicast IP addresses.



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| Setting | Description |
|----------------------|---|
| Name | Enter the name of the paging zone. Names can be a maximum of 15 characters. The paging zone name is displayed on the LCD during incoming and outgoing multicast pages. |
| Multicast IP | Enter the paging zone multicast IP address. The IP address range for multicast addresses is 224.0.0.0–239.255.255.255. |
| Multicast Port | Enter the multicast port used by the multicast IP address. The valid port range is 1 to 65535. |
| Priority | Select the paging zone priority from 1 to 10. Zones with a priority higher than another zone can interrupt the lower-priority zone's active page. In addition, a call priority setting is available in the configuration file (page_zone.call_priority_threshold). This priority setting also ranges from 1 to 10 (2 is the default). If the paging zone priority is higher or equal to the call priority, then a multicast page can interrupt an active, dialing, or incoming call. |
| Enable Incoming Page | Select to enable the VSP861 to receive incoming pages for that paging zone. If the "Enable Incoming Page" checkbox is not selected, the phone will not listen for the multicast, but will still be able to broadcast an outgoing page. |

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Server Application

On the Server Application page, you can enter Action URIs to allow the VSP861 to interact with a server application by using an HTTP GET request. The action URI triggers a GET request when a specified event occurs. Action URIs allow an external application to take control of the display when an event occurs. These pre-defined events are listed under "Action URI" on the Server Application page.

Action URIs are typically used in conjunction with the XML Browser, which can be customized to deliver an appropriate user experience.

The VSP861 supports both push and pull server applications. Note that Action URI events are not "push" events as it is the phone that requests a URI when triggered by certain states. You can enable push server applications under "XML Push Settings".

Action URI Syntax

To access an XML application, the phone performs an HTTP GET on a URL.

An HTTP GET request may contain a variable name and variable value, which are separated by "=". Each variable value starts and ends with "\$\$" in the query part of the URL.

Action URI variables pass dynamic data to the server. The valid URL format is: http://host[:port]/dir/file name?variable name=\$\$variable value\$\$

where:

- host is the hostname or IP address of the server supporting the XML application
- port is the port number the phones are using for the HTTP request

At the time of the HTTP call, the variable value field is populated with the appropriate data. For example, the following URL passes the SIP Account User Identifier to the server: http://10.50.10.140/script.pl?name=\$\$SIPUSERNAME\$\$

A GET request then passes along the following information:

http://10.50.10.140/script.pl?name=42512

Assuming that the User Identifier is 42512.

Variable names are defined by the particular XML application being called.

Variable values are predefined and depend on the status of the phone. If the variable has no meaning in the current status, then the phone sends an empty string.

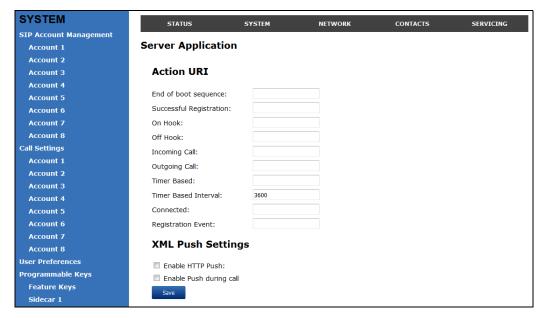
The table below lists all possible variable values. Note that variables applicable during an Incoming or Active Call (such as INCOMINGNAME and REMOTENUMBER) are initialized at the beginning and at the end of the call.

| Variable value | Description |
|----------------|-----------------------------|
| SIPUSERNAME | SIP Account User Identifier |
| DISPLAYNAME | SIP Account Display Name |

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| Variable value | Description |
|-------------------|---|
| LOCALIP | Phone's local IP Address |
| INCOMINGNAME | Caller ID name of the current Incoming Call |
| REMOTENUMBER | Remote party phone number (Incoming or Outgoing) |
| REGISTRATIONSTATE | Registration state available from the Registration event. Values are: REGISTERED DEREGISTERED FAIL |
| MAC | The phone's MAC Address |
| MODEL | The phone's model number: VSP861. |



Action URI

| Setting | Description |
|-------------------------|--|
| End of boot sequence | The End of boot sequence URI is triggered at the end of the phone boot sequence. Using the End of boot sequence URI, it is possible to develop self-provisioning on the phone. For example, an XML application can identify the phone and generate a MAC-specific file on the fly. |
| Successful Registration | The Successful Registration URI is triggered the first time the phone registers successfully to a SIP Account. If the phone registers to multiple SIP Accounts, then the Successful Registration URI is triggered for each line. |

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| Setting | Description |
|----------------------|---|
| On Hook | The On Hook URI is triggered when the phone transitions from Active to Idle (or from Paging to Idle). For example, when: |
| | The user presses the End button |
| | The user hangs up the corded handset during a call |
| | A transfer is completed and the user returns to idle |
| | ■ The far end hangs up |
| | The call was not answered |
| | ■ The call fails. |
| Off Hook | The Off Hook URI is triggered when the user goes to Dial mode by: |
| | Lifting the corded handset of the cradle |
| | Pressing the SPEAKER or HEADSET hard key |
| | Pressing a Line PFK |
| | Pressing the [New] button during a held call. |
| | Note that the Off Hook URI will NOT be triggered when calling a pre-defined number and going immediately to Dialing mode—this event triggers the Outgoing Call URI instead. |
| Incoming Call | The Incoming Call URI is triggered for each Incoming Ring event or Call Waiting event. Using the Incoming Call URI, it is possible to display extra information on the phone for an Incoming Call. For example, the XML application that is called when there is an Incoming Call can do a database lookup and display information on the caller. Note that this Action URI will not be triggered if DND or Call Forward All is enabled or if Call Waiting is disabled (i.e., the call is rejected). |
| Outgoing Call | The Outgoing Call URI is triggered each time a SIP INVITE message is sent (Dialing mode). For example, after: |
| | Pressing the Dial key in Pre-Dial with populated number |
| | Using the dial pad to speed dial a call |
| | Pressing a Quick Dial PFK |
| | Dialing a Directory number by going off-hook. |
| Timer Based | The Timer Based URI will be triggered when the configured timeout expires. The timer starts at the end of the phone boot sequence. |
| Timer Based Interval | Enter the interveral before the Timer Based URI is triggered. |
| - | |

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| Setting | Description |
|--------------------|---|
| Connected | The Connected URI is triggered each time the phone is in an Active Call or is Paging. |
| Registration Event | The Registration Event URI is triggered every time there is a registration state change. For example: |
| | Registered |
| | Deregistered |
| | Fail (Registration timed out, refused, or expired) |
| | The Registration Event URI is not triggered when the same event is repeated. |

XML Push Settings

| Setting | Description |
|-------------------------|--|
| Enable HTTP Push | Select to enable HTTP push, which enables the phone to display XML objects that are "pushed" to the phone from the server via http/https POST or SIP NOTIFY. |
| Enable Push during call | Select to enable the phone to display pushed XML objects during a call. Otherwise, the XML application is displayed after the call is over. |

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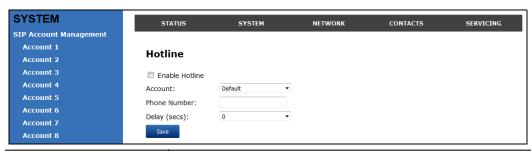
Hotline Settings

Hotline settings enable the VSP861 to dial a pre-configured number after any off-hook action in idle mode—lifting the handset, pressing **SPEAKER**, pressing **HEADSET**, or pressing a PFK Line key. The hotline dialing is subject to a delay. When this delay is configured, it supercedes the inter-digit timeout used for regular calls.

Predial mode and dialing mode are not considered idle mode. Hotline dialing will not trigger if digits have been entered, or when the phone is dialing a number.

Hotline dialing will be cancelled if the user presses any keys, or if there are any incoming calls during the hotline dialing delay period.

The hotline number uses the phone's default dial plan.



| Setting | Description |
|----------------|---|
| Enable Hotline | Enables the hotline feature. |
| Account | Sets the account used for dialing the hotline number. |
| Phone Number | Sets the number to be dialed after the hotline delay. |
| Delay (secs) | Sets the hotline delay before the number is dialed. |

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Local Call Recording

You can enable or disable local call recording and the call record tone on the Local Call Recording page. You can also delete recordings or export call recordings to the computer.



Before enabling call recording on the Local Call Recording page, an SD card must be inserted into the slot on the rear of the deskset.



| Setting | Description |
|-------------------------------|---|
| Enable Call Recording | Enable call recording for end users. |
| Enable Call Recording Tone | Enable a call recording tone that plays during the call when recording begins and ends. |

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Network Pages

You can set up the VSP861 for your network configuration on the Network pages. Your service provider may require you to configure your network to be compatible with its service, and the VSP861 settings must match the network settings.

The network settings are grouped into Basic and Advanced Settings. IPv4 and IPv6 protocols are supported.

When both IPv4 and IPv6 are enabled and available, the following guidelines apply when determining which stack to use:

- For outgoing traffic, the IP address (or resolved IP) in the server field—either IPv4 or IPv6—will determine which stack to be used.
- In general, most operations can be associated with one of the servers listed on the "Basic Network Settings" page. However, for operations triggered by/dependent upon network status, the phone must determine which server to use. For example, a special case like the "Network down" icon on the Deskset screen can be ambiguous for server association. Because its primary purpose is to aid in troubleshooting SIP registration issues, this case will be associated with the SIP registration server.
- DNS entries with both IPv4 and IPv6 settings can be used to resolve FQDN entries.
 There are no preferences with the order of the DNS queries.
- Pcap should include traffic for both stacks.
- Dual stack operations should be transparent to PC port traffic.



- PnP is not supported on IPv6.
- **NOTE** VPN is not supported in IPv6 or PPPoE.

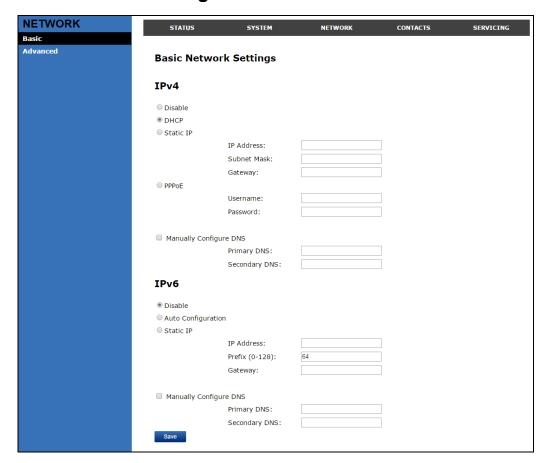
The network settings are also available as parameters in the configuration file. See "network Module: Network Settings" on page 158.

After entering information on this page, click Save to save it.

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Basic Network Settings





You must be familiar with TCP/IP principles and protocols to configure static IP NOTE settings.

Basic Network Settings

Click the link for each setting to see the matching configuration file parameter in "network Module: Network Settings" on page 158. Default values and ranges are listed there.

IPv4

| Setting | Description |
|---------|--|
| Disable | Disables all related IPv4 settings. |
| DHCP | DHCP is selected (enabled) by default, which means the VSP861 will get its IP address, Subnet Mask, Gateway, and DNS Server(s) from the network. When DHCP is disabled, you must enter a static IP address for the VSP861, as well as addresses for the Subnet Mask, Gateway, and DNS Server(s). |

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| Setting | Description |
|------------------------|--|
| Static IP | When Static IP is selected, you must enter a static IP address for the VSP861, as well as addresses for the Subnet Mask, Gateway, and DNS Server(s). |
| IP Address | If DHCP is disabled, enter a static IP address for the VSP861. |
| Subnet Mask | Enter the subnet mask. |
| Gateway | Enter the address of the default gateway (in this case, your router). |
| PPPoE | Select to enable PPPoE (Point-to-Point Protocol over Ethernet) mode. |
| PPPoE Username | Enter your PPPoE account username. |
| PPPoE password | Enter your PPPoE account password. |
| Manually Configure DNS | Select to enable manual DNS configuration. |
| Primary DNS | If DHCP is disabled, enter addresses for the primary and secondary DNS servers. |
| Secondary DNS | |

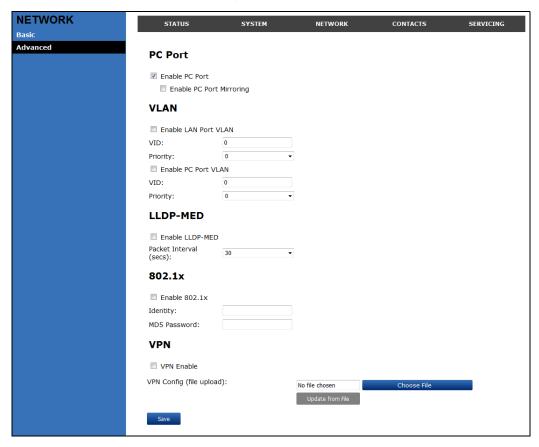
IPv6

| Setting | Description |
|------------------------|---|
| Disable | Disables all related IPv6 settings. |
| Auto Configuration | Auto configuration is selected (enabled) by default, which means the VSP861 will get its IP address, Gateway, and DNS Server(s) from the network. When Auto Configuration is disabled, you must enter a static IP address for the VSP861, as well as addresses for the Gateway and DNS Server(s). |
| Static IP | When Static IP is selected, you must enter a static IP address for the VSP861, as well as an IPv6 address prefix, Gateway, and DNS Server(s). |
| IP Address | If Auto Configuration is disabled, enter a static IP address for the VSP861. |
| Prefix (0-128) | Enter the IPv6 address prefix length (0 to 128 bits). |
| Gateway | Enter the address of the default gateway (in this case, your router). |
| Manually Configure DNS | Select to enable manual DNS configuration. |
| Primary DNS | If Auto Configuration is disabled, enter addresses for the primary and secondary DNS servers. |
| Secondary DNS | |

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Advanced Network Settings



PC Port

You can set the availability of the deskset PC port for network connectivity. When the port is enabled for connectivity, you can set the port to port mirroring, thereby allowing you to use the port to monitor inbound and outbound network traffic and facilitate troubleshooting.

| Setting | Description |
|--------------------------|---|
| Enable PC Port | Enable or disable the PC port to operate in hub/switch mode (depending on the Enable PC Port Mirroring setting). |
| Enable PC Port Mirroring | When the PC port is enabled, select Enable PC Port Mirroring to set the port to operate in hub mode (network traffic on the WAN port is reflected in the PC port). When Port Mirroring is not selected, the port operates in switch mode. |

VLAN

You can organize your network and optimize VoIP performance by creating a virtual LAN for phones and related devices.

Click the link for each setting to see the matching configuration file parameter in "network Module: Network Settings" on page 158. Default values and ranges are listed there.

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| Setting | Description |
|----------------------|---|
| Enable LAN Port VLAN | Enable if the phone is part of a VLAN on your network. Select to enable. |
| VID | Enter the VLAN ID (vlan 5, for example). |
| Priority | Select the VLAN priority that matches the Quality of Service (QOS) settings that you have set for that VLAN ID. Outbound SIP packets will be marked and sent according to their priority. 7 is the highest priority. Note: Configuring QOS settings for your router or switch is a subject outside the scope of this document. |
| Enable PC Port VLAN | Enable if the phone is part of a VLAN on your network. Select to enable. |
| VID | Enter the PC Port VLAN ID (vlan 5, for example). |
| Priority | Select the VLAN priority that matches the Quality of Service (QOS) settings that you have set for that VLAN ID. Outbound SIP packets will be marked and sent according to their priority. 7 is the highest priority. Note: Configuring QOS settings for your router or switch is a subject outside the scope of this document. |

LLDP-MED

| Setting | Description |
|------------------------|---|
| Enable LLDP-MED | Enables or disables Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED). LLDP-MED is a standards-based discovery protocol supported on some network switches. It is required for auto-configuration with VLAN settings. |
| Packet Interval (secs) | Sets the LLDP-MED packet interval (in seconds). |

802.1x

| Setting | Description |
|---------------|--|
| Enable 802.1x | Enables or disables the 802.1x authentication protocol. This protocol allows the phone to attach itself to network equipment that requires device authentication via 802.1x. |
| Identity | Enter the 802.1x EAPOL identity. |
| MD5 Password | Enter the 802.1x EAPOL MD5 password. |

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VPN

You can operate the VSP861 SIP deskset over a Virtual Private Network (VPN). VPN enables remote users and remote sites to connect to a main corporate network and SIP server with a high level of performance and security.

Configuring VPN using the WebUI consists of enabling VPN and uploading a VPN configuration file. The VPN configuration file (openvpn_client.tar) must contain the following files:

- client.conf
- a keys folder containing
 - ca.crt
 - client.crt
 - client.key

The filename of the VPN client configuration file and certificates must match the names provided above. For more information about configuring VPN, visit our website at businessphones.vtech.com or call 1 (888) 370-2006.



Ensure that NTP or manual time is configured correctly so that the VSP861 is using the correct date and time before VPN setup. Mismatched time between sites and servers may invalidate the initial TLS handshake.

| Setting | Description |
|--------------------------|---|
| VPN Enable | Enables or disables the phone to connect using the OpenVPN client. If VPN is enabled, but not connected, all SIP traffic will continue to route via the LAN IP. If VPN is enabled and connected, all SIP traffic will route via the VPN tunnel. The exception is the web server, which will still be accessible via the LAN IP. |
| VPN Config (file upload) | Browse to and upload the VPN configuration file openvpn_client.tar. |

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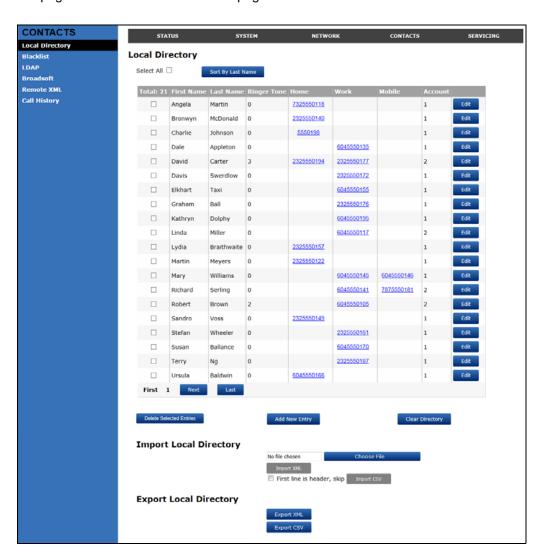


Contacts Pages

Local Directory

On the Local Directory page, you can manage your local directory entries. You can sort, edit, delete, and add contact information for up to 1000 entries. In order to back up your contacts or import another local directory file, the page also enables you to export and import your phone's local directory.

The Local Directory lists entries across multiple pages. Click Next, Last, First, or a page number to view the desired page of entries.





You can also use the phone menu to manage local directory entries. For more information, see the VSP861 User Guide.

Table 6 describes the buttons available on the Local Directory page.

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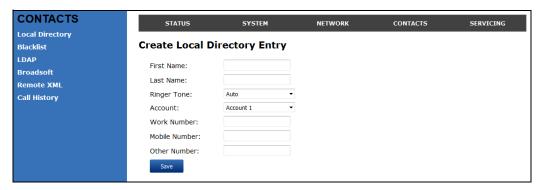


Table 6. Local Directory commands

| Click | То |
|-------------------------|--|
| Sort By Last Name | Sort the list by last name. |
| Edit | Edit information for an entry |
| Next | View the next page of entries. |
| Last | View the last page of entries. |
| First | View the first page of entries. |
| Delete Selected Entries | Delete selected entries from the directory. Click Select All to select every entry on the page you are viewing. |
| Add New Entry | Add a new directory entry. |
| Clear Directory | Delete all Directory entries. |
| Choose File | Import a directory file. |
| Export | Export the directory. |

To add a new directory entry:

Click Add New Entry .
 The Create Local Directory Entry page appears.



2. Enter the required information as described in the following table.

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Create Local Directory Entry

| Setting | Description | Range | Default |
|---------------|---|--------------------------------|--------------------|
| First Name | Enter the appropriate names in | | |
| Last Name | these fields. The maximum length of the first name and last name fields is 15 characters. | n/a | Blank |
| Ringer Tone | Sets a unique ringer tone for calls from this directory entry. | Auto, Tone 1–10 | Tone 1 |
| Account | Sets the account used when you dial this directory entry. | Default Account, Account 1– | Default Account |
| Work Number | | | |
| Mobile Number | Enter the appropriate names and numbers in these fields. | n/a | Blank |
| Other Number | | | |

Directory Import/Export

The best way to create a directory file for import is to first export the directory from the phone. The directory can be exported as an .xml or .csv file. After exporting the file, open it in an .xml or .csv editor and add or modify entries.



When importing a .csv file, you can select whether the first line should be treated as a header and ignored for the import.

Importing a directory file adds the imported directory entries to existing entries. Therefore, it is possible to have duplicate entries after importing a directory file. If you are importing a "complete" directory file with the aim of replacing the entire current directory, use **Select All** and Delete Selected Entries to clear the directory before importing the file.



Using the configuration file, you can set whether an imported directory file adds to existing entries or replaces existing entries. See *"file Module: Imported File Parameters"* on page 201.

Directory files in .xml format have the following tags:

| Local Directory WebUI field | Directory file XML tag |
|-----------------------------|--|
| First Name | <dir_entry_name_first></dir_entry_name_first> |
| Last Name | <dir_entry_name_last></dir_entry_name_last> |
| Work Number | <dir_entry_number_work></dir_entry_number_work> |
| Mobile Number | <pre><dir_entry_number_mobile></dir_entry_number_mobile></pre> |
| Other Number | <dir_entry_number_other></dir_entry_number_other> |
| Account | <dir_entry_line_number></dir_entry_line_number> |

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| Local Directory WebUI field | Directory file XML tag |
|-----------------------------|---------------------------------------|
| Call Block (not on WebUI) | <dir_entry_block></dir_entry_block> |
| Ringer Tone | <dir_entry_ringer></dir_entry_ringer> |

Blacklist

On the Blacklist page, you can manage local blacklist entries. The VSP861 rejects calls from numbers that match blacklist entries. You can sort, edit, delete, and add up to 1000 blacklist entries. In order to back up your blacklist entries or import another local blacklist file, the page also enables you to export and import the blacklist.

The blacklist lists entries across multple pages. Click Next, Last, First, or a page number to view the desired page of entries.



You can also use the VSP861 menu to manage blacklist entries. For more information, see the VSP861 User Guide.

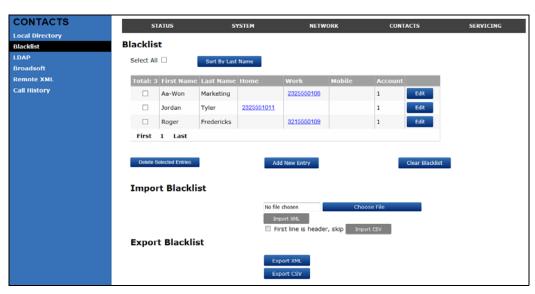


Table 7 describes the buttons available on the Blacklist page.

Table 7. Blacklist commands

| Click | То |
|-------------------|--------------------------------|
| Sort By Last Name | Sort the list by last name. |
| Edit | Edit information for an entry |
| Next | View the next page of entries. |
| Last | View the last page of entries. |

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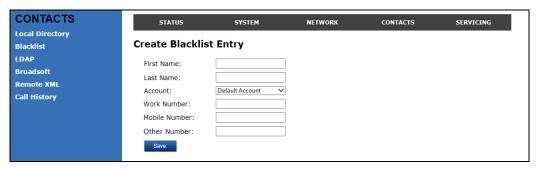


Table 7. Blacklist commands

| Click | То |
|-------------------------|---|
| First | View the first page of entries. |
| Delete Selected Entries | Delete selected entries. Click Select All to select every entry on the page you are viewing. |
| Add New Entry | Add a new entry. |
| Clear Directory | Delete all entries. |
| Choose File | Import a blacklist file. |
| Export | Export the blacklist. |

To add a new blacklist entry:

Click Add New Entry .
 The Create Blacklist Entry page appears.



2. Enter the required information as described in the following table.

Create Blacklist Entry

| Setting | Description | Range | Default |
|---------------|---|---------------------------------|-----------|
| First Name | Enter the appropriate names in | | |
| Last Name | these fields. The maximum length of the first name and last name fields is 15 characters. | n/a | Blank |
| Account | Sets the account used when you dial this directory entry. | Default Account, Account 1–8 | Account 1 |
| Work Number | | | |
| Mobile Number | Enter the appropriate names and numbers in these fields. | n/a | Blank |
| Other Number | | | |

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Blacklist Import/Export

The best way to create a blacklist file for import is to first export the blacklist from the VSP861. The blacklist can be exported as an .xml or .csv file. After exporting the file, open it in an .xml or .csv editor and add or modify entries.



When importing a .csv file, you can select whether the first line should be treated as a header and ignored for the import.

Importing a blacklist file adds the imported blacklist entries to existing entries. Therefore, it is possible to have duplicate entries after importing a blacklist file. If you are importing a "complete" blacklist file with the aim of replacing the entire current blacklist, use **Select All** and Delete Selected Entries to clear the blacklist before importing the file.



Using the configuration file, you can set whether an imported blacklist file adds to or replaces existing entries. See *"file Module: Imported File Parameters"* on page 201.

Blacklist files in .xml format have the following tags:

| Blacklist WebUI field | Blacklist file XML tag |
|-----------------------|---|
| First Name | <blacklist_entry_name_first></blacklist_entry_name_first> |
| Last Name | <blacklist_entry_name_last></blacklist_entry_name_last> |
| Work Number | <blacklist_entry_number_work></blacklist_entry_number_work> |
| Mobile Number | <blacklist_entry_number_mobile></blacklist_entry_number_mobile> |
| Other Number | <blacklist_entry_number_other></blacklist_entry_number_other> |
| Account | <blacklist_entry_line_number></blacklist_entry_line_number> |

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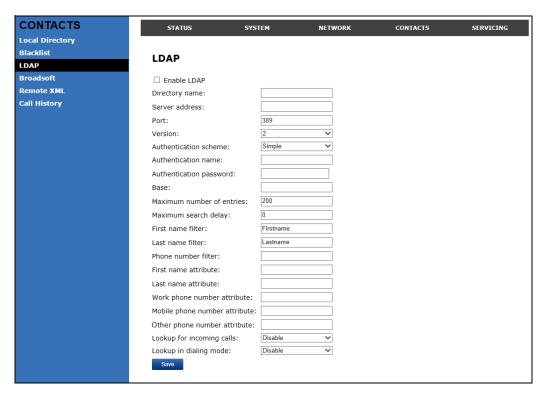


LDAP

The phone supports remote Lightweight Directory Access Protocol (LDAP) directories. An LDAP directory is hosted on a remote server and may be the central directory for a large organization spread across several cities, offices, and departments. You can configure the phone to access the directory and allow users to search the directory for names and telephone numbers.

The LDAP settings are also available as parameters in the configuration file. See *"remoteDir Module: Remote Directory Settings" on page 175.*

After entering information on this page, click Save to save it.



LDAP Settings

Click the link for each setting to see the matching configuration file parameter in "remoteDir Module: Remote Directory Settings" on page 175. Default values and ranges are listed there.

About LDAP attribute filters

The LDAP filters on this page give you control over how directory entry search results are determined. For example, consider if **gn** is the firstname attribute and **sn** is the lastname attribute in the LDAP search base. The filter <attribute>=% returns records based on the beginning of the user-entered string. If gn=% is used for a firstname filter, entering "da" returns records such as Daisy, Dale, David, etc.

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The filter <attribute>=* returns records containing the user-entered string anywhere in that attribute. If gn=* is used for a firstname filter, entering "ar" returns records such as Karen, Arnold, Gary, etc.

The filter (|(gn=%)(sn=%)) returns firstname and lastname records that start with the user-entered string.

LDAP number filters give you the same control over number searches and matches. If for example, you have defined the number attributes **telephoneNumber**, **mobile** and **otherPhone** for Work, Mobile and Other numbers respectively, then the filter (|(telephoneNumber=*)(mobile=*)(otherPhone=*)) will display the correct directory information if the number (from an incoming call, or a dialed number) matches a number in any three of those fields.

The filter telephoneNumber=* will display the correct directory information if the incoming call number matches a number in the "Work" field only.

| Setting | Description |
|---------------------------|--|
| Enable LDAP | Enables or disables the phone's access to the LDAP directory. |
| Directory name | Enter the LDAP directory name. |
| Server address | Enter the LDAP server domain name or IP address. |
| Port | Enter the LDAP server port. |
| Version | Select the LDAP protocol version supported on the phone. Ensure the protocol value matches the version assigned on the LDAP server. |
| Authentication scheme | Select the LDAP server authentication scheme. |
| Authentication name | Enter the user name or authentication name for LDAP server access. |
| Authentication password | Enter the authentication password for LDAP server access. |
| Base | Enter the LDAP search base. This sets where the search begins in the directory tree structure. Enter one of more attribute definitions or LDAP field names, separated by commas (no spaces). Your directory may include attributes like "cn" (common name) or "ou" (organizational unit) or "dc" (domain component). For example: ou=accounting,dc=vtech,dc=com |
| Maximum number of entries | Sets the maximum number of entries returned for an LDAP search. Limiting the number of hits can conserve network bandwidth. |
| Maximum search delay | Enter the delay (in seconds) before the phone starts returning search results. |

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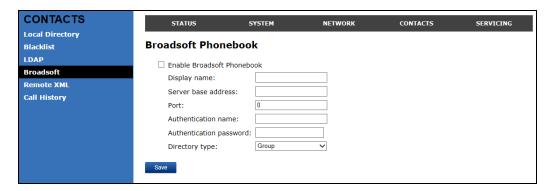
| Setting | Description |
|---------------------------|---|
| First name filter | Enter the first name attributes for LDAP searching. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). |
| Last name filter | Enter the last name attributes for LDAP searching. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). |
| Phone number filter | Enter the number attributes for LDAP searching. The format of the search filter is compliant to the standard string representations of LDAP search filters (RFC 2254). |
| First name attribute | Sets the attribute for first name. What you enter here should match the first name attribute for entries on the LDAP server (gn for givenName, for example). This helps ensure that the phone displays LDAP entries in the same format as the Local Directory. |
| Last name attribute | Sets the attribute for last name. What you enter here should match the last name attribute for entries on the LDAP server (sn for surname, for example). This helps ensure that the phone displays LDAP entries in the same format as the Local Directory. |
| Work number attribute | Sets the attribute for the work number. What you enter here should match the work number attribute for entries on the LDAP server (telephoneNumber, for example). This helps ensure that the phone displays LDAP entries in the same format as the Local Directory. |
| Mobile number attribute | Sets the attribute for the mobile number. What you enter here should match the mobile number attribute for entries on the LDAP server (mobile, for example). This helps ensure that the phone displays LDAP entries in the same format as the Local Directory. |
| Other number attribute | Sets the attribute for the other number. What you enter here should match the other number attribute for entries on the LDAP server (otherPhone, for example). This helps ensure that the phone displays LDAP entries in the same format as the Local Directory. |
| Lookup for incoming calls | Enables or disables LDAP incoming call lookup. If enabled, the phone searches the LDAP directory for the incoming call number. If the number is found, the phone uses the LDAP entry for CID info. |
| Lookup in dialing mode | Enables or disables LDAP outgoing call lookup. If enabled, numbers entered in pre-dial or live dial are matched against LDAP entries. If a match is found, the LDAP entry is displayed for dialing. |

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Broadsoft

The phone supports access to the Broadsoft Phonebook. Users can search for and call contacts that are hosted on the Broadsoft Phonebook. On the Broadsoft Phonebook Settings page, you must enter the path and credentials to enable the phone to access the Broadsoft Phonebook.



Broadsoft Phonebook Settings

| Setting | Description |
|-------------------------------|---|
| Enable Broadsoft Phonebook | Enables or disables the phone's access to the Broadsoft phonebook. |
| Display name | Enter the display name for the Broadsoft Phonebook. This name appears on the Directory list on the VSP861 menu. |
| Server base address | Enter the Broadsoft Phonebook server domain or IP address. |
| Port | Enter the Broadsoft Phonebook server port. |
| Authentication name | Enter the user name or authentication name for Broadsoft Phonebook access. |
| Authentication password | Enter the authentication password for Broadsoft Phonebook access. |
| Directory type | Select the directory type: Group, Group Common, Enterprise, Enterprise Common, Personal |

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Remote XML

The VSP861 supports three server-hosted Remote XML directories. A total of 5000 Remote XML directory entries are supported. The 5000 entries can be shared across the three remote XML directories.

When the user selects a remote directory to view, the VSP861 will sync with the directory server. The phone will display **Sync failed**. if any of the following failing conditions is encountered:

- Server not reachable
- Remote XML directory file is not available
- Invalid XML directory file

Remote XML Directory Format

The following shows a sample single-entry file which can be used in a remote XML directory. Note that the default tags are the same as those defined for the Local Directory.

```
<?xml version="1.0" encoding="utf-8"?>

<DIR_ENTRY>

<DIR_ENTRY_NAME_FIRST>John</DIR_ENTRY_NAME_FIRST>

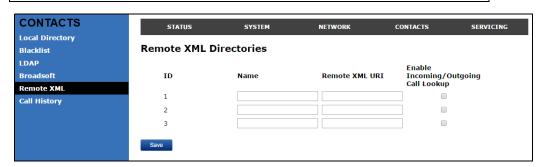
<DIR_ENTRY_NAME_LAST>Smith</DIR_ENTRY_NAME_LAST>

<DIR_ENTRY_NUMBER_OTHER>33333</DIR_ENTRY_NUMBER_OTHER>

<DIR_ENTRY_NUMBER_WORK>1111</DIR_ENTRY_NUMBER_WORK>

<DIR_ENTRY_NUMBER_MOBILE>2222</DIR_ENTRY_NUMBER_MOBILE>

</DIR_ENTRY>
```



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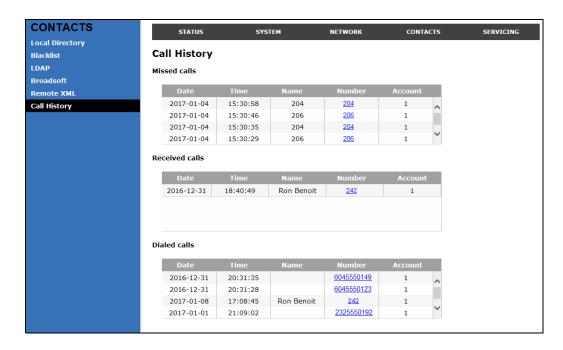
| Setting | Description |
|--|---|
| Name | Sets the name of the directory as it will appear on the VSP861 Directory list. The following order applies to the Directory list when multiple server-based directories are enabled: 1. Local 2. Blacklist 3. LDAP 4. Broadsoft 5. Remote XML directory 1 6. Remote XML directory 2 7. Remote XML directory 3 Any Remote XML directories will move up the list if LDAP and/or Broadsoft directories are not enabled. |
| Remote XML URI | Enter the location of the XML directory file, from which the phone will sync and retrieve directory entries. |
| Enable Incoming/ Outgoing Call Lookup | Enables/disables the call lookup feature for incoming and outgoing calls. |

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Call History

The Call History page has no configurable settings. It displays Missed Calls, Received Calls, and Dialed Calls. Users can view their call history and "click to dial" numbers if click to dial is enabled.



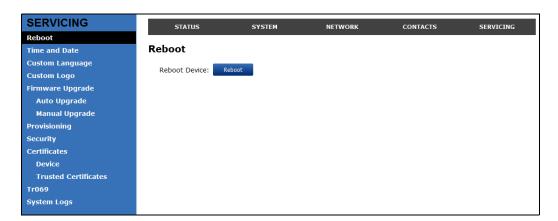
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Servicing Pages

Reboot

To manually reboot the VSP861 and apply settings that you have updated, click Reboot



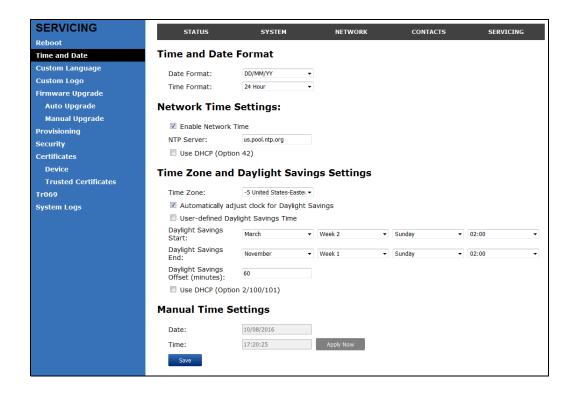
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Time and Date

On the Time and Date page, you can manually set the time and date, and the time and date formats. You can also set the system time to follow a Network Time Protocol (NTP) Server (recommended) or you can set the time and date manually.

The time and date settings are also available as parameters in the configuration file. See "time_date Module: Time and Date Settings" on page 170.



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Time and Date Format

Click the link for each setting to see the matching configuration file parameter in "time_date Module: Time and Date Settings" on page 170. Default values and ranges are listed there.

| Setting | Description |
|-------------|--|
| Date Format | Sets the date format. |
| Time Format | Sets the clock to a 24-hour or 12-hour format. |

Network Time Settings

| Setting | Description |
|----------------------|--|
| Enable Network Time | Enables or disables getting time and date information for your phone from the Internet. |
| NTP Server | If Enable Network Time is selected, enter the URL of your preferred time server. |
| Use DHCP (Option 42) | If Enable Network Time is selected, select to use DHCP to locate the time server. Option 42 specifies the NTP server available to the phone. When enabled, the phone obtains the time in the following priority: 1. Option 42 2. NTP Server 3. Manual time. |

Time Zone and Daylight Savings Settings

| Setting | Description |
|--|---|
| Time Zone | Select your time zone from the list. |
| Automatically adjust clock for Daylight Savings | Select to adjust the clock for daylight savings time according to the NTP server and time zone setting. To disable daylight savings adjustment, disable both this setting and User-defined Daylight Savings Time. |
| User-defined Daylight Savings Time | Select to set your own start and end dates and offset for Daylight Savings Time. To disable daylight savings adjustment, disable both this setting and Automatically adjust clock for Daylight Savings. |
| DST Start: Month DST Start: Week DST Start: Day DST Start: Hour | If User-defined DST is enabled, set the start date and time for daylight savings: Month, week, day, and hour. |
| DST End: Month DST End: Week DST End: Day DST End: Hour | If User-defined DST is enabled, set the end date and time for daylight savings: Month, week, day, and hour. |

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| Setting | Description |
|-----------------------------|---|
| Daylight Savings Offset | If User-defined DST is enabled, this specifies the daylight savings adjustment (in minutes) to be applied when the current time is between Daylight Savings Start and Daylight Savings End. |
| Use DHCP (Option 2/100/101) | If Enable Network Time is selected, select to use DHCP to determine the time zone offset. Options 2, 100 and 101 determine time zone information. |

Manual Time Settings

If Enable Network Time is disabled or if the time server is not available, use Manual Time Settings to set the current time.

| Setting | Description |
|---------|--|
| Date | Select the current year, month, and day. Click the Date field and select the date from the calendar that appears. |
| Time | Sets the current hour, minute, and second. Click the Time field, and enter the current time. You can also refresh the page to update the manual time settings. |

Click Apply Now to start the VSP861 using the manual time settings.

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Custom Language

On the Export Translation page, you can export WebUI and/or phone language strings. After exporting language strings, you can use the resulting file as the basis for a custom language translation file (.tpk file).

You can import one custom language for use on the device user interface and the WebUI. The custom language adds to the existing languages available with the firmware. Separate translation files are required for the device user interface and the WebUI.

Importing a custom language can only be done using the configuration file. See *"file Module: Imported File Parameters"* on page 201. For details on creating and formatting a translation file, see the Custom Language document at *businessphones.vtech.com*.



The available languages for export are identical to the WebUI and Phone Language lists described in "User Preferences" on page 69.

The filenames of the exported language files will be:

- WebUI: <Model Number>-<Display Name>-webui.tpk
- Device MMI: <Model Number>-<Display Name>-phoneui.tpk

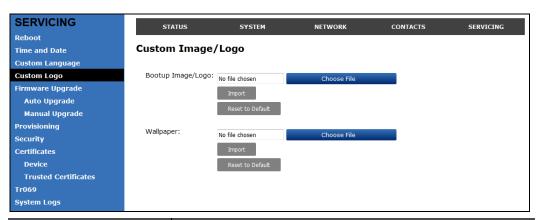
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Custom Image/Logo

You can upload a custom image or logo to be displayed on the VSP861 idle screen and during bootup. For more information about formatting a custom logo file, see "Adding a Custom Logo" on page 21.

On this page, you can also reset the bootup and idle logos to factory defaults. The default logo for bootup and idle mode is the **vtech** logo.



| Setting | Description |
|-------------------|---|
| Bootup Image/Logo | Import a custom image or logo shown during bootup. For logo specifications, see "Logo specifications" on page 21. |
| Wallpaper | Import an image that will appear as the background of the Home screen. |

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Firmware Upgrade

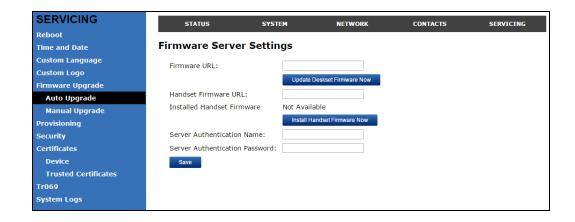
You can update the VSP861 with new firmware using the following methods:

- Retrieving a firmware update file from a remote host computer and accessed via a URL. This central location may be arranged by you, an authorized dealer, or your SIP service provider. Enter the URL under Firmware Server Settings.
- Using a file located on your computer or local network. No connection to the Internet is required. Consult your dealer for access to firmware update files. Click Manual Upgrade to view the page where you can manually upgrade the VSP861 firmware.

The firmware upgrade settings are also available as parameters in the configuration file. See "provisioning Module: Provisioning Settings" on page 163.

Firmware Server Settings

Click the link for each setting to see the matching configuration file parameter in *"provisioning Module: Provisioning Settings"* on page 163. Default values and ranges are listed there.



| Setting | Description |
|--------------------------------|---|
| Firmware URL | The URL where the firmware update file resides. This should be a full path, including the filename of the firmware file. |
| Handset Firmware URL | The URL where the Cordless Handset Accessory firmware update file resides. This should be a full path, including the filename of the firmware file. |
| Server authentication name | Authentication username for the firmware server |
| Server authentication password | Authentication password for the firmware server |

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To update the firmware immediately:

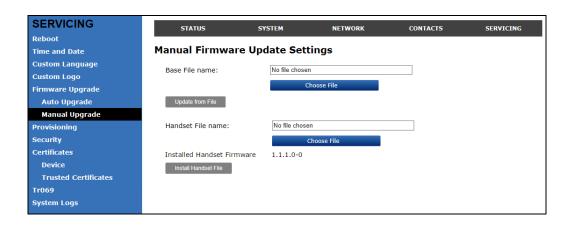
■ Click Update Deskset Firmware Now Or Install Handset Firmware Now



You can also configure the VSP861 to check for firmware updates at regular intervals. See "Provisioning" on page 114.

Manual Firmware Update and Upload

On the Manual Firmware Update Settings page, you can upgrade the VSP861 firmware using a file located on your computer or local network.



To update the firmware using a file on your computer or local network:

- 1. On the Manual Firmware Update page, click Choose File to locate and open the firmware update file.
- 2. Click Update from File or Install Handset File

After clicking Update from File the VSP861 will update its firmware and restart. If you are updating handset firmware, you must perform one more step after clicking Install Handset File

Updating a Cordless Handset

Updating DECT cordless handset firmware using the WebUI is a two-step process. First you must download the handset firmware and install it on the deskset. Second, you must install the handset firmware on the handset. The handset downloads the firmware over the air from the deskset.

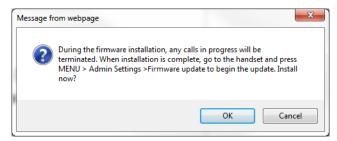
To install the handset firmware on the deskset:

Click Install Handset Firmware Now Manual Firmware update.
 for the Firmware Server update or Install Handset File for the Manual Firmware update.

The confirmation dialog box shown below appears.

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2. To install the handset firmware, click ok . The message Installing handset firmware. Please wait... appears. To cancel the download, click cancel .

After clicking ok , the message **System update in progress**. **Please wait**... appears on the handset.

After a successful update, the message **Firmware installation successful** appears on the WebUI.

An error message appears if:

- the handset firmware is aleady up to date.
- the handset firmware URL is incorrect, or the file cannot be retrieved for any other reason.
- the handset firmware file is corrupted.
- the handset doesn't recognize the firmware file. For example, the firmware file may belong to a different ErisTerminal product.

To install the firmware on the cordless handset:

- 1. On the handset, press **MENU**, and then select **Admin settings**.
- 2. Enter the admin password. The default is **admin**. To switch between entering upper or lower-case letters, press the * key.
- On the Admin settings menu, select Firmware update.
 The handset checks for new firmware. If new firmware is found, the handset screen asks you to proceed with the update.

Provisioning

Provisioning refers to the process of acquiring and applying new settings for the VSP861 using configuration files retrieved from a remote computer. After a VSP861 is deployed, subsequent provisioning can update the VSP861 with new settings; for example, if your service provider releases new features. See also "Provisioning Using Configuration Files" on page 132.

With automatic provisioning, you enable the VSP861 to get its settings automatically—the process occurs in the background as part of routine system operation. Automatic provisioning can apply to multiple devices simultaneously.

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With manual provisioning on the WebUI, you update the VSP861 settings (configuration and/or firmware) yourself via **Provisioning > Import Configuration** and/or **Firmware Upgrade > Manual Upgrade**. Manual provisioning can only be performed on one VSP861 at a time.

On the Provisioning page, you can enter settings that will enable the VSP861 to receive automatic configuration and firmware updates. The Provisioning page also allows you to manually update VSP861 configuration from a locally stored configuration file using an Import function. You can also export the VSP861 configuration—either to back it up or apply the configuration to another VSP861 in the future—to a file on your computer.

The provisioning process functions according to the Resynchronization settings and Provisioning Server Settings. The VSP861 checks for the provisioning URL from the following sources in the order listed below:

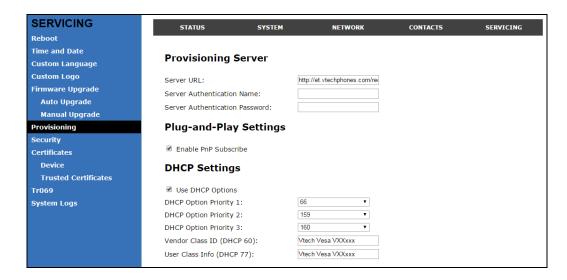
- 1. PnP—Plug and Play Subscribe and Notify protocol
- 2. DHCP Options
- 3. Preconfigured URL—Any VSP861 updated to the latest firmware release will have the Redirection Server URL available as the default Provisioning Server URL (see "provisioning.server address" on page 163).



Using the Redirection Service requires contacting the VTech support team for an account.

If one of these sources is disabled, not available, or has not been configured, the VSP861 proceeds to the next source until reaching the end of the list.

The provisioning settings are also available as parameters in the configuration file. See *"provisioning Module: Provisioning Settings"* on page 163.



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Provisioning Settings

| Setting | Description |
|--------------------------------|---|
| Server URL | URL of the provisioning file(s). The format of the URL must be RFC 1738 compliant, as follows: " <schema>://<user>:<password>@ <host>:<port>/<url-path>" "<user>:<password>@" may be empty. "<port>" can be omitted if you do not need to specify the port number. The default URL is the VTech redirect server: https://et.vtechphones.com/rg2/</port></password></user></url-path></port></host></password></user></schema> |
| Server authentication name | User name for access to the provisioning server |
| Server authentication password | Password for access to the provisioning server |

Plug-and-Play Settings

| Setting | Description |
|----------------------|---|
| Enable PnP Subscribe | Select to enable the VSP861 to search for the provisioning URL via a SUBSCRIBE message to a multicast address (224.0.1.75). The VSP861 expects the server to reply with a NOTIFY that includes the provisioning URL. The process times out after five attempts. |

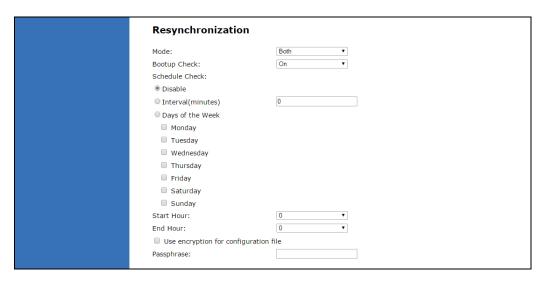
DHCP Settings

| Setting | Description |
|---------------------------|---|
| Use DHCP Options | Enables the VSP861 to use DHCP options to locate and retrieve the configuration file. When selected, the VSP861 automatically attempts to get a provisioning server address, and then the configuration file. If DHCP options do not locate a configuration file, then the server provisioning string is checked. Note: Ensure that DHCP is also enabled on the "Basic Network Settings" page. |
| DHCP Option Priority 1 | If DHCP is enabled, sets the DHCP Option priority. Select the highest priority option. |
| DHCP Option Priority 2 | If DHCP is enabled, sets the DHCP Option priority. Select the second highest priority option. |
| DHCP Option Priority 3 | If DHCP is enabled, sets the DHCP Option priority. Select the third highest priority option. |
| Vendor Class ID (DHCP 60) | DHCP Option 60 is available to send vendor-specific information to the DHCP Server. |

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| Setting | Description |
|---------------------------|---|
| User Class Info (DHCP 77) | DHCP Option 77 is available to send vendor-specific information to the DHCP Server. |



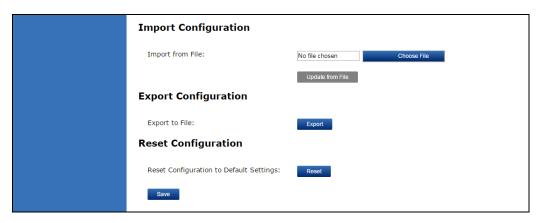
Resynchronization

| Setting | Description |
|-------------------------------------|--|
| Mode | Sets which files for which the VSP861 checks. It can check for configuration files, firmware update files (from the URL entered on the Firmware Server Settings page), or both. Note: When checking for both configuration and firmware files, the firmware URL can be within the config file. This firmware URL takes take precedence over the URL on the Firmware Server Settings page. It will also update the URL on the Firmware Server Settings page. This allows you to change the firmware URL automatically. |
| Bootup Check | Sets the VSP861 to check the provisioning URL for new configuration and/or firmware files upon bootup. The update is applied as part of the reboot process. |
| Schedule Check: Disable | When selected, disables regularly scheduled file checking. |
| Schedule Check: Interval | Sets an interval for checking for updates. After selecting Interval, enter the interval in minutes between update checks. |
| Schedule Check: Days of the Week | Select to enable weekly checking for updates on one or more days. After selecting Days of the Week, select the day(s) on which the VSP861 checks for updates. |
| Start Hour | Select the hour of the day on which the VSP861 checks for updates. |
| End Hour | Select the hour of the day on which the VSP861 stops checking for updates. |

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| Setting | Description |
|----------------|---|
| Use encryption | Enables an AES-encrypted configuration file to be decrypted before being applied to the VSP861. Select if the configuration file has been secured using AES encryption. See "Securing configuration files with AES encryption" on page 138. |
| Passphrase | If the configuration file has been secured using AES encryption, enter the 16-bit key. See "Securing configuration files with AES encryption" on page 138. |



Import Configuration

You can configure the VSP861 by importing a configuration file from your computer or your local network. For more information about configuration file types and configuration file formatting, see "Provisioning Using Configuration Files" on page 132.

To import a configuration file:

- Click Choose File to locate and open the configuration file.
- 2. Click Update from File .

The VSP861 will update its configuration.

Manually importing a configuration file differs from the auto-provisioning process in that:

- The VSP861 does not check whether the file has been loaded before. The configuration file is processed whether or not it is different from the current version.
- The VSP861 will restart immediately after importing the configuration file, without waiting for one minute of inactivity.

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Export Configuration

You can export all the settings you have configured on the WebUI and save them as a configuration file on your computer. You can then use this configuration file as a backup, or use it to update other phones.

Under Export Configuration, you can also reset the phone to its default configuration.



The exported configuration file will contain null values for the following passwords:

- SIP account authentication password
- EAPOL password
- PPPoE password
- Firmware server password
- Provisioning server password
- Encryption passphrase
- TR-069 password
- TR-069 connection request password
- Administrator access password
- User access password
- LDAP server password
- Broadsoft directory server password.

Please ensure that you save the exported configuration file in a secure location. You can also enable passwords to be exported as plain text.

See "provisioning.pwd export enable" on page 167.

To export the configuration file:

■ Click Export .

The format of the exported file is <model name>_<mac address>.cfg. For example, VSP861 0011A0OCF489.cfg.

Exporting a configuration file generates two header lines in the configuration file. These header lines provide the model number and software version in the following format:

#Model Number = xxxxxxx

#SW Version = xxxxxxx

You can use the exported file as a general configuration file, and duplicate the settings across multiple units. However, ensure that you edit the file to remove any MAC-specific SIP account settings before applying the general configuration file to other units.



Reset Configuration

You can reset the phone to its default settings.

To reset the VSP861 to its default configuration:

- 1. Under **Reset Configuration**, click Reset .
- 2. When the confirmation box appears, click **OK**.

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Security

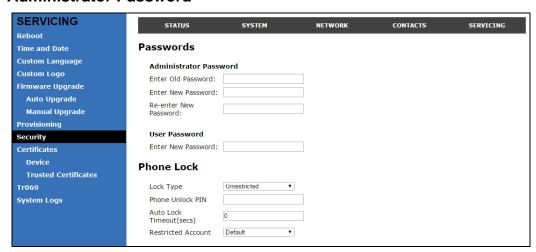
On the **Security** page you can reset the admin password, reset the user password, configure the phone lock feature, and enter web server settings.



By default, after bootup, the VSP861 alerts you if the default passwords (user password and admin password) are still in use.

The security settings are also available as parameters in the configuration file. See "web Module: Web Settings" on page 181.

Administrator Password



You can set the administrator password on the WebUI or by using provisioning. For more information on using provisioning to set the administrator password, see "profile Module: Password Settings" on page 216.

To change the admin password:

- 1. Enter the old password (for a new VSP861, the default password is **admin**).
- 2. Enter and re-enter a new password. The password is case sensitive and can consist of both numbers and letters (to a maximum of 15 characters).
- 3. Click Save .

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User Password

You can set the user password on the WebUI or by using provisioning. For more information on using provisioning to set the user password, see "profile Module: Password Settings" on page 216.

To change the User password:

- 1. Enter the old password (for a new VSP861, the default password is **user**).
- 2. Enter and re-enter a new password. The password is case sensitive and can consist of both numbers and letters (to a maximum of 15 characters).
- 3. Click Save .

Phone Lock

The Phone Lock feature restricts certain hard keys and features unless the user enters a PIN code. For more information about the phone lock feature, see "Using the Phone Lock menu" on page 41.

| Setting | Description |
|--------------------------|--|
| Lock Type | Unrestricted, Emergency Call Only. |
| Phone Unlock PIN | Enter the PIN that the user enters to unlock the phone. |
| Auto Lock Timeout (secs) | Enter a timeout period in seconds. When the phone becomes idle after being used in unlocked mode, the phone automatically locks after the timeout. |
| Restricted Account | Select the account to be used when Phone Lock is active. |

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PIN Masking

The PIN masking feature allows users to hide PIN numbers during call operation. The different types of PIN masking are:

- Hide DTMF digits—when Hide DTMF is enabled, PIN numbers entered during an Active Call are automatically masked.
- Password Dial—when Password Dial is enabled, PIN numbers entered as part of the dial string are hidden according to the Password Dial Prefix and Password Dial Length settings described below.

PIN numbers in a dial string are masked in Pre-Dial and Dial modes (including variants of these modes, such as Transfer Setup, Conference Setup, and Incoming Call Forward Setup).



Masked PIN numbers are not saved in Redial entries. The masked numbers are stripped out of the dial string before being saved in the Call History.

| Setting | Description |
|-------------------------------------|---|
| Enable Hidden DTMF digits | Enable to mask all DTMF digits entered during an Active Call. Note that unlike Password Dial, there is no prefix parameter and no length parameter, so any DTMF string of any length is masked. |
| Enable Delay for Hidden DTMF digits | Enable to add a one-second delay before DTMF digits are masked. By default, the digits are masked immediately. |
| Enable Password Dial | Enable to hide PIN numbers entered as part of the dial string. |
| Enable Delay for Password Dial | Enable to add a one-second delay before PIN numbers are masked. By default, the PIN number is masked immediately. |
| Password Dial Prefix | Enter the prefix that serves as an indicator that the next x digits are masked (x being equal to the Password Dial Length). Only the first matched Prefix initiates PIN masking. If additional instances of the Password Dial Prefix appear elsewhere within the dial string, they are ignored. |
| Password Dial Length | Enter the PIN number length. All digits within this length are masked. Any digits beyond this length are not masked. For example, if the Password Prefix is 99 and the Password Dial Length is 3 , then 9912345 will be rendered as 99***45 . |

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Web Server



| Setting | Description |
|------------------------|--|
| HTTP Server port | Port used by the HTTP server. |
| Enable Secure Browsing | Sets the server to use the HTTPS protocol. |
| HTTPS Server port | Port used by the HTTPS server. |

To configure Web Server Settings:

- 1. Enter the HTTP Server port number. The default setting is 80.
- 2. Enable or Disable Secure Browsing. When enabled, the HTTPS protocol is used, and you must select the HTTPS server port in the next step.
- 3. Enter the HTTPS server port number. The default setting is 443.



Changing the Web Server settings will reboot the VSP861.

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Trusted Servers

The Trusted Servers setting provides a means of blocking unauthorized SIP traffic. When enabled, each account's Registration server, SIP server, Outbound Proxy server and Backup Outbound Proxy server will be used as sources for trusted SIP traffic. All unsolicited SIP traffic (for example, INVITE, NOTIFY, unsolicited MWI, OPTIONS) will be blocked unless it is from one of the trusted servers with the enabled accounts.

If additional trusted sources are required beyond what has been specified with the enabled accounts (for example, if IP dialing or other types of server traffic need to be secured), use the Trusted IP settings on the Security page.

| Trusted Servers |
|---------------------------------|
| Accept SIP account servers only |

| Setting | Description |
|---------------------------------|---|
| Accept SIP account servers only | Enable or disable using the account servers as sources for trusted SIP traffic. |

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Trusted IP

In addition to the Trusted Servers setting, incoming IP traffic can be filtered using an "Allowed IP" list of IP addresses. When this means is enabled, all unsolicited IP traffic will be blocked unless it is from one of the trusted IP addresses on the "Allowed IP" list.

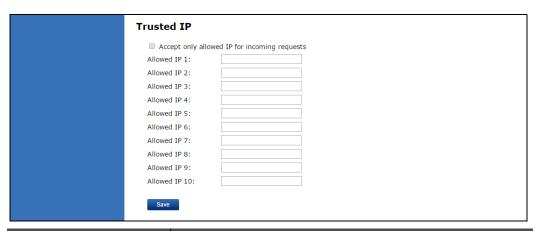
Yu can enter the "Allowed IP" list in the 10 fields on the "Trusted IP" section. Entries on the "Allowed IP" list must be specified as IP addresses (IPv4 or IPv6).

Three formats are supported for entries on the "Allowed IP" list:

- 1. IP range specified using CIDR notation (defined in rfc4632). IPv4 or IPv6 address followed by a prefix; for example, 192.168.0.1/24.
- 2. IP range specified with a pair of starting and ending IPv4 or IPv6 addresses, separated by '-' (for example, 192.168.0.1-192.168.5.6).
 - No space before or after '-'
 - Both starting IP & ending IP have to be with the same IP version
 - Starting IP has to be smaller than the ending IP; otherwise, all traffic will be dropped.
- 3. Single IP address in IPv4 or IPv6.



To ensure WebUI access after configuring Trusted IP, you must include the IP of the Web Browser on the "Allowed IP" list.



| Setting | Description |
|--|--|
| Accept only allowed IP for incoming requests | Enable or disable using the "Allowed IP" list to filter all IP traffic. |
| Allowed IP 1–10 | Enter IP addresses or address ranges to be used as sources of authorized IP traffic. |

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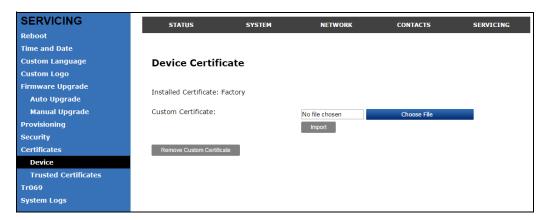


Certificates

You can add two types of certificates using the WebUI or the provisioning file (see "file Module: Imported File Parameters" on page 201). The two types of certificates are:

- Device—A single Device Certificate can be uploaded so that other parties can authenticate the phone in the following cases:
 - When the phone acts as a web server for the user to manage configurations.
 - When the phone acts as a client for applications where HTTP is supported.
- Trusted—Trusted Certificates are for server authentication with secured HTTP transaction in the following applications: SIP signaling, Provisioning, Firmware, LDAP directory service, and Broadsoft directory service. Up to 20 trusted certificates can be installed.

Device Certificate



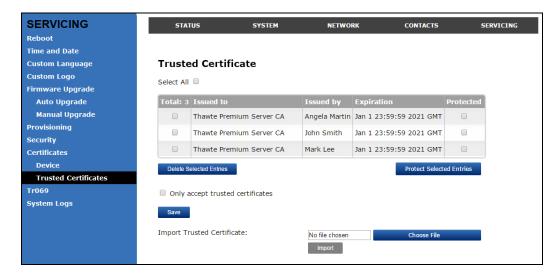
To upload a device certificate:

- 1. On the Device Certificate page, click Choose File .
- 2. Locate the certificate file and click **Open**.
- 3. On the Device Certificate page, click Import.

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Trusted Certificate



On the Trusted Certificate page, you can:

- import up to 20 trusted certificates.
- delete individual (or all) certificates.
- protect certificates by selecting them in the **Protected** column, and then clicking Protect Selected Entries . Protected certificates cannot be selected for deletion and are not removed during a reset to factory defaults.

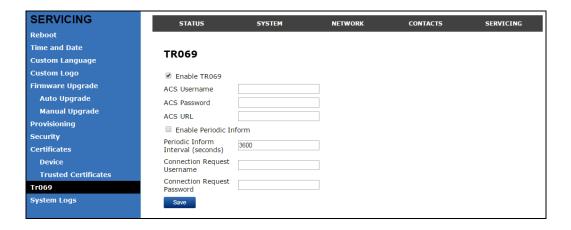
Select **Only accept trusted certificates** to enable server authentication. Deselecting this option disables server authentication.

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TR-069 Settings

The Broadband Forum's Technical Report 069 (TR-069) defines a protocol for remote management and secure auto-configuration of compatible devices. On the TR069 page, you can enable TR-069 and configure access to an auto-configuration server (ACS).



| Setting | Description |
|------------------------------------|---|
| Enable TR069 | Enable/Disable TR-069 subsystem. |
| ACS Username | User name used for ACS authentication. |
| ACS Password | Password used for ACS authentication. |
| ACS URL | URL used to contact the ACS (for example, http://my.acs:9675/path/to/somewhere/). |
| Enable Period Inform | Enable/Disable periodic inform method calls. |
| Periodic Inform Interval (seconds) | Periodic inform method calls interval. |
| Connection Request Username | If the ACS wants to communicate with the device, it must offer the matching Connection Request user name. When the device sends the report to ACS for the first time, it contains information for this. |
| Connection Request Password | If the ACS wants to communicate with the device, it must offer the matching Connection Request password. When the device sends the report to ACS for the first time, it contains information for this. |

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System Logs

On the **Syslog Settings** page, you can enter settings related to system logging activities. It supports the following logging modes:

- Syslog server
- Volatile file

Under **Network Trace**, you can capture network traffic related to the phone's activity and save the capture as a .pcap file. The file can be used for diagnostic and troubleshooting purposes.

Under **Download Log**, you can save the system log to a file.

The Syslog settings are also available as parameters in the configuration file. See "log Module: System Log Settings" on page 174.



Syslog Settings

| Setting | Description |
|----------------|-------------------------------------|
| Enable Syslog | Enable log output to syslog server. |
| Server address | Syslog server IP address. |
| Server port | Syslog server port. |

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| Setting | Description |
|-----------|--|
| Log Level | Sets the log level. The higher the level, the larger the debug output. |
| | ■ 5—ALL |
| | ■ 4—DEBUG |
| | ■ 3—INFO |
| | ■ 2—WARNING |
| | ■ 1—ERROR |
| | ■ 0—CRITICAL |

The logging levels are:

- CRITICAL: Operating conditions to be reported or corrected immediately (for example, an internal component failure or file system error).
- ERROR: Non-urgent failures—unexpected conditions that won't cause the device to malfunction.
- WARNING: An indication that an error or critical condition can occur if action is not taken.
- INFO: Normal operational messages.
- DEBUG: Developer messages for troubleshooting/debugging purposes.

Network Trace

To perform a network trace:

- 1. Start a network trace by clicking Start . The button changes to Stop .
- 2. Stop the network trace by clicking Stop .
- 3. Save the trace by clicking Save to file. Your browser should prompt you to save the capture.pcap file.

Download Log

To download the system log:

- 1. Click Save Log to file .
- 2. After your browser prompts you to save the **system.log** file, save the file in the desired location.

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CHAPTER 4

Provisioning Using Configuration Files

Provisioning using configuration files is the quickest way to configure multiple VSP861 desksets. You can place configuration files on a provisioning server, where the VSP861 desksets retrieve the files and update their configuration automatically.

Configuration files have the extension **.cfg** and contain settings that will apply to VSP861 desksets. To edit a configuration file, open it with a text editor such as Notepad.

The settings within a configuration file are grouped into modules. Most of the modules group their settings in the same way that settings are grouped on the VSP861 WebUI. For example, the "time_date" module in the configuration file contains the same settings that are on the **Time and Date** WebUI page. For a complete list of VSP861 configuration file modules and their associated parameters, see "Configuration File Parameter Guide" on page 140.

Using the WebUI, you can also import a configuration file and apply the configuration file settings to the VSP861. For more information, see "Import Configuration" on page 118.

This chapter covers:

- "The Provisioning Process" on page 133
- "Configuration File Types" on page 135
- "Data Files" on page 136
- "Configuration File Tips and Security" on page 137.



The Provisioning Process

The automatic provisioning process is as follows:

Check for new or updated configuration files. For file-checking options, see
 "Provisioning" on page 114 and "Resynchronization: configuration file checking" on
 page 134. The VSP861 maintains a list of the last loaded provisioning files. The
 VSP861 compares its current configuration against the files it finds on the provisioning
 server. Checking for update... appears on the VSP861 screen.

If provisioning has been triggered by the resync timer expiring or by remote check-sync, the VSP861 checks for updated files after one minute of inactivity.

2. Download the configuration files.

If any file on the provisioning server has changed, the VSP861 treats it as a new file and downloads it. **Configuring Deskset...** appears on the VSP861 screen.

If the provisioning URL specifies a path only with no filename (if the URL ends with "/"), then by default the VSP861 looks for and retrieves the following two files by appending the two default filenames to the URL:

- General file: <model>.cfg.
- MAC-specific file: <model>_<MAC Address>.cfg.

The <model> variable is the VTech product model: VSP861, for example.

If the provisioning URL contains a query element (?), or a filename ending in ".cfg" is specified at the end of the provided URL path, then the VSP861 retrieves only the configuration file specified.

 The VSP861 restarts after one minute of inactivity. Please wait while the phone reboots appears on the VSP861 screen. For more information, see "VSP861 restart" on page 134.

During provisioning, the VSP861 reads the configuration file and validates each module and setting. The VSP861 considers a setting valid if it is:

- a valid data type
- formatted as a valid setting
- within a valid data range
- part of a module that passes an integrity check. That is, the module's settings are consistent and logical. For example, in the "network" module, if DHCP is disabled, but no static IP address is specified, the module will fail the integrity check and none of the settings will apply.

Invalid modules or invalid settings are skipped and logged as ERROR messages in the system log, but will not interrupt the provisioning process. The system log will include the module parameters that have not been applied. A recognized module with unrecognized settings will cause all other settings in that module to be skipped.



A successful configuration or firmware update is reported as an INFO message in the system log.

See "Configuration File Parameter Guide" on page 140 for the options and value ranges available for each configuration file setting.

Resynchronization: configuration file checking

You can select a number of options that determine when the VSP861 checks for new configuration files. This process of checking for configuration files is called Resynchronization. Resynchronization options are available on the WebUI **Provisioning** page, but you can also include them in a configuration file.

The resynchronization options are:

- Mode—sets the VSP861 to check for a configuration file only, a firmware update file only, or both types of file.
- Never—configuration file checking is disabled
- Bootup—the VSP861 checks for new configuration files when it boots up. Any updates are applied during the boot-up process.
- Remote check-sync—enables you to start a resynchronization remotely using your hosted server's web portal. The Remote check-sync settings are available only in the configuration file, not the WebUI.
- Repeatedly, at a defined interval from 60 to 65535 minutes (45 days).

VSP861 restart

If the VSP861 needs to restart after an auto-update, the restart happens only after the device has been idle for one minute.

To prevent users from delaying the update process (auto-updates cannot begin until the VSP861 has been idle for one minute), or to avoid device restarts that might interfere with incoming calls:

- set the resynchronization interval to a suitable period
- upload any new configuration file(s) to your provisioning server after work hours so that the VSP861 will download the file(s) when there is no call activity.

When you update the VSP861 by importing a configuration file using the WebUI, the device restarts immediately after applying the new settings, regardless of whether the VSP861 is idle.



Configuration File Types

The VSP861 is able to retrieve and download two types of configuration file. Depending on your requirements, you may want to make both types of configuration file available on your provisioning server.

The two configuration file types are a general configuration file and a MAC-specific configuration file. The types differ in name only. The formatting of the files' content is the same.

The general configuration file contains settings that are required by every VSP861 in the system.

The MAC-specific configuration file is a file that only a single VSP861 can retrieve. The MAC-specific configuration file name contains a VSP861 MAC address and can only be retrieved by the device with a matching MAC address.

The filename formats for both files are:

- General file: <model>.cfg
- MAC-specific file: <model>_<MAC Address>.cfg

The <model> variable is the VTech product model; for example, **VSP861**. For more information about the MAC-specific configuration file, see "Guidelines for the MAC-Specific configuration file" on page 137.

Both the general and MAC-specific files can contain any of the available configuration settings. A setting can appear in the general configuration file or the MAC-specific configuration file, or both files, or neither file. If a setting appears in both files, the setting that is read last is the one that applies.

When the VSP861 fetches both a general and a MAC-specific configuration file, the general file is processed first. You can configure a setting for most of your VSP861 desksets in the general file, and then overwrite that setting for just a few VSP861 desksets using the MAC-specific file.



Data Files

The configuration file can also include links to data files for product customization. Allowed data types include the following:

- Directory (contacts, blacklist) in .xml format
- Certificates (server, provisioning, LDAP, Broadsoft) in pem format
- Logos (a bootup logo and an idle screen logo) in .bmp format

Links to data files are in the configuration file's "file" module. This is where you enter any URLs to the data files that the VSP861 deskset may require.

None of the data files are exported when you export a configuration file from the VSP861. However, you can export a Directory or Blacklist .xml file using the WebUI. After modifying the .xml file, you can use the configuration file "file" module to have the VSP861 import the new file. For a complete list of data file parameters, see *"file Module: Imported File Parameters" on page 201*.



Configuration File Tips and Security

All configuration settings are initially stored in a configuration template file. Copy, rename, and edit the template file to create a general configuration file and the MAC-specific configuration files you will need. You can store the general configuration file and the MAC-specific files on your provisioning server.

Do not modify the configuration file header line that includes the model and firmware version.

To save yourself time and effort, consider which settings will be common to all (or the majority of) VSP861 desksets. Such settings might include call settings, language, and NAT settings. You can then edit those settings in the configuration template and save it as the general configuration file. The remaining settings will make up the MAC-specific configuration file, which you will have to copy and edit for each VSP861.

Clearing parameters with %NULL in configuration file

For configuration file parameters that can have a text string value, you can clear the value of the parameter by applying the value %NULL in the configuration file.

For example: sip_account.1.display_name = %NULL

However, the following parameters are exceptions. Applying the value %NULL to these parameters will reset them to their default value.

- file.bootup_logo applying %NULL restores the default value (VTech logo)
- file.idle_logo applying %NULL restores the default value (VTech logo)
- file.custom_ringer applying %NULL restores the default value

Guidelines for the MAC-Specific configuration file

The VSP861 downloads the MAC-specific configuration file after the general configuration file. You must create a MAC-specific configuration file for each VSP861 in your system. The file name must contain the VSP861 MAC address, which is printed on a label on the back of the device, or available on the **MENU > Status > Product Info** screen. For example, a VTech VSP861 deskset with the MAC address of 00:11:A0:10:6F:2D would download the **VSP861_0011A0106F2D.cfg** file.



When renaming a MAC-specific configuration file, ensure the filename is all upper case.

The MAC-specific configuration file contains settings intended exclusively for that VSP861 deskset. Such settings will include SIP account settings such as display name, user ID, and authentication ID.



Securing configuration files with AES encryption

You can encrypt your configuration files to prevent unauthorized users modifying the configuration files. The VSP861 firmware decrypts files using the AES 256 algorithm. After encrypting a file and placing it on your provisioning server, you can enable the VSP861 to decrypt the file after fetching it from the server.

The procedures in this section use OpenSSL for Windows for file encryption, as shown in Figure 2.

To decrypt a configuration file, you will need a 16-character AES key that you specified when you encrypted the file. The key (or passphrase) is limited to 16 characters in length and supports special characters $\sim ^ ` \% ! \& - _ + = | . @ * : ; , ? () [] {} <> / \# as well as spaces.$



The encryption of configuration files is supported only for the auto provisioning process. Encrypt files only if you intend to store them on a provisioning server. Do not encrypt files that you intend to manually import to the VSP861. You cannot enable decryption for manually imported configuration files.

To encrypt a configuration file:

- 1. (Optional) Place your configuration file in the same folder as the opensal executable file. If the configuration file is not in the same folder as the opensal executable file, you can enter a relative pathname for the [infile] in the next step.
- 2. Double-click the **openssl.exe** file.
- 3. On the openssl command line, type:

```
enc -aes-256-cbc -pass pass: [passphrase123456] -in [infile] -out [outfile] -nosalt -p
```

Elements in brackets are examples—do not enter the brackets. Enter a 16-character passphrase and the unencrypted configuration file filename (the "infile") and a name for the encrypted file ("outfile") that will result.

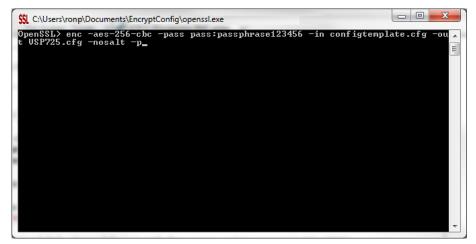
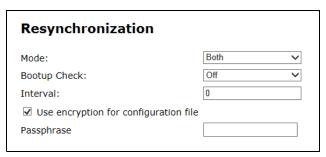


Figure 2. OpenSSL command line



To enable configuration file decryption:

- 1. On the WebUI, click **Servicing > Provisioning**.
- 2. On the Provisioning page under **Resynchronization**, select **Use Encryption for configuration file**.



- 3. Enter the 16-character passphrase that you created when you encrypted the configuration file.
- 4. Click Save .



You must ensure that configuration files are encrypted when enabling AES Encryption. Decrypting an unencrypted file will result in a garbage file that is not processed. This will also be logged as an error in the system log.



CHAPTER 5

CONFIGURATION FILE PARAMETER GUIDE

This chapter lists the available options for all the settings within the VSP861 configuration file. Most settings in the configuration file have an equivalent in the WebUI (see the settings tables in "Using the WebUI" on page 45). However, the options you must enter when editing the configuration file have a different syntax and format.

The settings are divided into modules. Most modules correspond to a page on the VSP861 WebUI. You may wish to reorganize the modules within the configuration file itself. The configuration file settings can be listed in any order, and the configuration file will still be valid.

The modules included in the configuration file are:

- "sip_account Module: SIP Account Settings" on page 141
- "hs_settings Module: Handset Settings" on page 157
- "network Module: Network Settings" on page 158
- "provisioning Module: Provisioning Settings" on page 163
- "security Module: Security Settings" on page 168
- "time_date Module: Time and Date Settings" on page 170
- "log Module: System Log Settings" on page 174
- "remoteDir Module: Remote Directory Settings" on page 175
- "web Module: Web Settings" on page 181
- "trusted_ip Module: Trusted IP Settings" on page 182
- "trusted_servers Module: Trusted Server Settings" on page 183
- "user_pref Module: User Preference Settings" on page 184



- "call_settings Module: Call Settings" on page 188
- "pfk Module: Programmable Feature Key Settings" on page 192
- "speed_dial Module: Speed Dial Settings" on page 196
- "audio Module: Audio Settings" on page 197
- "ringersetting Module: Distinctive Ringer Settings" on page 199
- "call_record Module: Call Recording Settings" on page 200
- "file Module: Imported File Parameters" on page 201
- "xml_app Module: XML App Settings" on page 205
- "system_event Module: Action URI Settings" on page 206
- "tr069 Module: TR-069 Settings" on page 208
- "tone Module: Tone Definition Settings" on page 210
- "profile Module: Password Settings" on page 216
- "page_zone Module: Paging Zone Settings" on page 218
- "phonelock Module: Phone Lock Settings" on page 220

sip_account Module: SIP Account Settings

The SIP Account settings enable you to set up individual accounts for each user. Each account requires you to configure the same group of SIP account settings. The SIP account settings for each account are identified by the account number, from 1 to for the VSP861.

For example, for account 1 you would set:

```
sip_account.1.sip_account_enable = 1
sip_account.1.label = Line 1
sip_account.1.display_name = 1001
sip_account.1.user_id = 2325551001
and so on.
```

For account 2, you would set:

```
sip_account.2.sip_account_enable = 1
sip_account.2.label = Line 2
sip_account.2.display_name = 1002
sip_account.2.user_id = 2325551002
```

and so on, if you have additional accounts to configure.



The SIP account settings follow the format: sip_account.x.[element], where x is an account number ranging from 1 to for the VSP861.

All these settings are exported when you manually export the configuration from the VSP861.

General configuration file settings

Setting: sip_account.x.dial_plan

Description: Sets the dial plan for account x. See "Dial Plan" on page 54.

Values: Text string Default: x+P

Setting: sip_account.x.call_restrict_dial_plan

Description: Sets the call restriction dial plan, which prevents users from completing

calls to certain numbers for this account.

Values: Text string Default: blank

Setting: sip_account.x.emergency_dial_plan

Description: Sets the emergency dial plan.

Values: Text string Default: blank

Setting: sip_account.x.inter_digit_timeout

Description: Sets the inter-digit timeout (in seconds) for account x. The inter-digit

timeout sets how long the VSP861 waits after the last digit is entered

before dialing the number.

Values: 1–10 Default: 3

Setting: sip_account.x.maximum_call_number

Description: Sets the maximum number of concurrent active calls allowed for that

account.

Values: 1- Default:

Setting: sip account.x.auto answer enable

Description: Enables or disables automatic answering of pages for account x.

Values: 0 (disabled), 1 (enabled) Default: 0



Setting: sip account.x.auto answer during active call

Description: Enables or disables automatic answering of pages for account x when

account x has an active call.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.barge in enable

Description: If the shared line type is enabled for account x, enables or disables

"barge in" capability for VSP861 desksets with shared accounts.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.dtmf transport method

Description: Sets the transport method for DTMF signaling for account x.

Values: auto, rfc2833, inband, info Default: auto

Setting: sip account.x.unregister after reboot enable

Description: Enables or disables the VSP861 to unregister account x after rebooting.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip_account.x.primary_sip_server_address

Description: Sets the SIP server IP address for account x.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: sip_account.x.primary_sip_server_port

Description: Sets the SIP server port for account x.

Values: 1–65535 **Default:** 5060

Setting: sip_account.x.primary_registration_server_address

Description: Sets the registration server IP address for account x.

Values: IPv4, IPv6 or FQDN Default: Blank



Setting: sip account.x.primary registration server port

Description: Sets the registration server port for account x.

Values: 1–65535 **Default:** 5060

Setting: sip_account.x.primary_registration_expires

Description: Sets the expiration time (in seconds) of the current registration for

account x.

Values: 30–7200 **Default:** 3600

Setting: sip account.x.registration retry time

Description: Sets the retry frequency of the current registration for account x.

Values: 1–1800 **Default:** 10

Setting: sip account.x.primary outbound proxy server address

Description: Sets the outbound proxy server IP address for account x.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: sip account.x.primary outbound proxy server port

Description: Sets the outbound proxy server port for account x.

Values: 1–65535 **Default:** 5060

Setting: sip account.x.backup outbound proxy server address

Description: Sets the backup outbound proxy server IP address for account x.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: sip_account.x.backup_outbound_proxy_server_port

Description: Sets the backup outbound proxy server port for account x.

Values: 1–65535 **Default:** 5060



Setting: sip_account.x.codec_priority.1

Description: Sets the highest-priority codec for account x.

Values: g711u, g711a, g729, g726, **Default:** g711u

g722, g723_1, ilbc

Setting: sip account.x.codec priority.2

Description: Sets the second highest-priority codec for account x.

Values: none, g711u, g711a, g729, Default: g711a

g726, g722, g723_1, ilbc

Setting: sip_account.x.codec_priority.3

Description: Sets the third highest-priority codec for account x.

Values: none, g711u, g711a, g729, Default: g726

g726, g722, g723_1, ilbc

Setting: sip_account.x.codec_priority.4

Description: Sets the fourth highest-priority codec for account x.

Values: none, g711u, g711a, g729, **Default:** g722

g726, g722, g723_1, ilbc

Setting: sip_account.x.codec_priority.5

Description: Sets the fifth highest-priority codec for account x.

Values: none, g711u, g711a, g729, Default: ilbc

g726, g722, g723_1, ilbc

Setting: sip_account.x.codec_priority.6

Description: Sets the sixth highest-priority codec for account x.

Values: none, g711u, g711a, g729, **Default:** g723_1

g726, g722, g723_1, ilbc

Setting: sip_account.x.codec_priority.7

Description: Sets the lowest-priority codec for account x.

Values: none, g711u, g711a, g729, **Default:** g729

g726, g722, g723_1, ilbc



Setting: sip_account.x.voice_encryption_enable

Description: Enables or disables SRTP voice encryption for account x.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip_account.x.g729_annexb_enable

Description: Enables G.729 Annex B, with voice activity detection (VAD) and

bandwidth-conserving silence suppression. This setting applies only

when G.729 is selected in a sip_account.x.codec_priority

parameter.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.dscp

Description: Sets the Voice Quality of Service Layer 3 - DSCP for account x.

Values: 0–63 **Default:** 46

Setting: sip_account.x.sip_dscp

Description: Sets the Signaling Quality of Service Layer 3 - DSCP for account x.

Values: 0–63 **Default:** 26

Setting: sip_account.x.local_sip_port

Description: Sets the Local SIP port for account x.

Values: 1–65535 **Default:** Account 1: 5060

Account 2: 5070 Account 3: 5080 Account 4: 5090

Setting: sip_account.x.transport_mode

Description: Sets the Signaling Transport Mode for account x.

Values: udp, tcp, tls Default: udp



Setting: sip account.x.blf variant

Description: Sets the BLF operation for account x. This parameter is not available on

the WebUI.

"default" is for Broadsoft or Asterisk depending on the presence of

sip_account.x.blf_list_uri.
"avaya" is designed as an Avaya variant.

"extended_blf" is proprietary.

"metaswitch" is designated for Metaswitch. "freeswitch" is designated for Freeswitch.

Values: default, avaya, Default: default

extended_blf, metaswitch,

freeswitch

Setting: sip_account.x.blf_subscription_expires

Description: Sets the BLF subscription expiry time (in seconds) for account x.

Values: 15–65535 **Default:** 3600

Setting: sip account.x.blf remote pickup code

Description: Sets the Busy Lamp Field (BLF) remote pickup code for account x.

Values: Text string Default: Blank

Setting: sip account.x.mwi enable

Description: Enables or disables message waiting indicator subscription for account x.

Enable if SUBSCRIBE and NOTIFY methods are used for MWI.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.mwi subscription expires

Description: Sets the MWI subscription expiry time (in seconds) for account x.

Values: 15–65535 **Default:** 3600

Setting: sip account.x.mwi ignore unsolicited

Description: Enables or disables ignoring of unsolicited MWI notifications—

notifications in addition to, or instead of, SUBSCRIBE and NOTIFY methods—for account x. Disable if MWI service is configured on the voicemail server and does not involve a subscription to a voicemail server.



Setting: sip_account.x.stutter_dial_tone_enable

Description: Enables or disables MWI stutter dial tone for account x.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: sip_account.x.nat_traversal_stun_enable

Description: Enables or disables STUN (Simple Traversal of UDP through NATs) for

account x. STUN enables clients, each behind a firewall, to establish calls via a service provider hosted outside of either local network.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.nat traversal stun server address

Description: Sets the STUN server IP address.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: sip_account.x.nat_traversal_stun_server_port

Description: Sets the STUN server port.

Values: 1–65535 **Default:** 3478

Setting: sip_account.x.nat_traversal_stun_keep_alive_enable

Description: Enables or disables STUN keep-alives. Keep-alive packets are used to

maintain connections established through NAT.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: sip_account.x.nat_traversal_stun_keep_alive_interval

Description: Sets the interval (in seconds) for sending keep-alives.

Values: 0–65535 **Default:** 30

Setting: sip account.x.keep alive enable

Description: Enable SIP keep alive in service of NAT traversal and as a heartbeat

mechanism to audit the SIP server health status.



Setting: sip account.x.keep alive interval

Description: Sets the interval (in seconds) for sending keep-alives.

Values: 1–3600 **Default:** 15

Setting: sip_account.x.keep_alive_ignore_failure

Description: Enable the phone to ignore keep-alive failure, if the failure can trigger

account re-registration and re-subscription (and active calls are dropped).

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: sip account.x.music on hold enable

Description: Enables or disables a hold-reminder tone that a far-end caller hears

when put on hold during a call on account x.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: sip_account.x.sip_session_timer_enable

Description: Enables or disables the SIP session timer.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.sip session timer min

Description: Sets the session timer minimum value (in seconds) for account x.

Values: 90–65535 **Default:** 90

Setting: sip_account.x.sip_session_timer_max

Description: Sets the session timer maximum value (in seconds) for account x.

Values: 90–65535 **Default:** 1800

Setting: sip account.x.check trusted certificate

Description: Enables or disables accepting only a trusted TLS certificate for account x.



Setting: sip account.use first trusted certificate for all

Description: Enables or disables accepting the first TLS certificate for all accounts.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.park variant

Description: Selects how the VSP861 handles call parking, depending on the service

provider. When the default "broadsoft" variant is selected, calls are parked via a PFK and a feature access code. The target number for the "park" request is formed by concatenating "Call Park FAC" of the call-to-be-parked account and the value entered for the Park PFK. When the "asterisk" variant is selected, calls are parked through a blind transfer to a parking lot extension. The target parking lot extension will be

taken from the following order of priority (if both values exist):

Park PFK Value

Call Park FAC Value

Values: broadsoft, asterisk Default: broadsoft

Setting: sip account.x.server side ctrl variant

Description: Set the server type that will control feature sync and FAC operation.

Values: default, comverse Default: default

Setting: sip account.x.preferred ptime

Description: Enter the packetization interval time in milliseconds.

Values: 10, 20, 30, 40, 50, 60 **Default:** 20

Setting: sip_account.x.cid_src priority.1

Description: Set the desired caller ID source to be displayed on the incoming call

screen.

Values: pai, rpid, from **Default:** pai

Setting: sip account.x.cid src priority.2

Description: Select the lower-priority caller ID source.

Values: pai, rpid, from Default: rpid



Setting: sip account.x.cid src priority.3

Description: Select the lowest-priority caller ID source.

Values: pai, rpid, from Default: from

Setting: sip_account.x.call_rejection_response_code

Description: Select the response code for call rejection. This code applies to the

following call rejection cases:

■ User presses Reject for an incoming call (except when Call

Forward Busy is enabled)

DND is enabled

Phone rejects a second incoming call with Call Waiting disabled

Phone rejects an anonymous call with Anonymous Call Rejection

enabled

Phone rejects call when the maximum number of calls is reached

Values: 480, 486, 603 **Default:** 486

Setting: sip_account.x.dtmf_payload_type

Description: Set the configurable RTP payload type for in-call DTMF.

Values: 96–127 **Default:** 101

MAC-specific configuration file settings

Setting: sip account.x.sip account enable

Description: Enables account x to be used by the device.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip account.x.label

Description: Sets the text that identifies the account on the device LCD. The account

label appears on the idle screen, dialing screen, and other call

appearance screens.



Setting: sip_account.x.display_name

Description: Sets the text portion of the caller ID that is displayed for outgoing calls

using account x.

Values: Text string Default: Blank

Setting: sip account.x.user id

Description: Sets the account ID for account x. Depending on your service provider's

specifications, this could be an extension number.

Note: Do not enter the host name (e.g. "@sipservice.com"). The configuration file automatically adds the default host name.

Values: Text string Default: Blank

Setting: sip_account.x.authentication_name

Description: Sets the authentication name for account x. Depending on your service

provider's specifications, this could be identical to the user ID.

Values: Text string Default: Blank

Setting: sip account.x.authentication access password

Description: Sets the authentication password for account x.

Values: Text string Default: Blank

Setting: sip_account.x.feature_sync_enable

Description: Enables or disables feature synchronization for account x. When

enabled, features configured on the service provider's web portal will

automatically be updated on the device's WebUI.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip_account.x.share_line_enable

Description: Sets the account type for account x. If the shared line type is enabled,

multiple VSP861 desksets can be configured with shared line

appearances.



Setting: sip account.x.access code page

Description: Sets the paging feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_park_call

Description: Sets the Call Park feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code retrieve parked call

Description: Sets the retrieve parked call feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code retrieve voicemail

Description: Sets the voicemail retrieval feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code dnd on

Description: Sets the do not disturb (DND) ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code dnd off

Description: Sets the do not disturb (DND) OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_cfa_on

Description: Sets the Call Forward All ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_cfa_off

Description: Sets the Call Forward All OFF feature access code for account x.



Setting: sip account.x.access code cfna on

Description: Sets the Call Forward No Answer ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_cfna_off

Description: Sets the Call Forward No Answer OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code cfb on

Description: Sets the Call Forward Busy ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_cfb_off

Description: Sets the Call Forward Busy OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code anonymous call block on

Description: Sets the Anonymous Call Block ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code anonymous call block off

Description: Sets the Anonymous Call Block OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_outgoing_call_anonymous_on

Description: Sets the Anonymous Outgoing Call ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_outgoing_call_anonymous_off

Description: Sets the Anonymous Outgoing Call OFF feature access code for account x.



Setting: sip account.x.access code call waiting on

Description: Sets the Call Waiting ON feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_code_call_waiting_off

Description: Sets the Call Waiting OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code group call pickup

Description: Sets the Group Call Pickup feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access code direct call pickup

Description: Sets the Direct Call Pickup feature access code for account x.

Values: Text string Default: Blank

Setting: sip account.x.access sf on

Description: Sets the Comverse Secretarial Filtering ON feature access code for

account x.

Values: Text string Default: Blank

Setting: sip account.x.access sf off

Description: Sets the Comverse Secretarial Filtering OFF feature access code for

account x.

Values: Text string Default: Blank

Setting: sip_account.x.access_hg_on

Description: Sets the Comverse Hunt Group ON feature access code for account x.



Setting: sip_account.x.access_hg_off

Description: Sets the Comverse Hunt Group OFF feature access code for account x.

Values: Text string Default: Blank

Setting: sip_account.x.blf_list_uri

Description: Sets the Busy Lamp Field (BLF) list URI for account x. The device will

retrieve the list from this location.

Values: SIP URI text string Default: Blank

Setting: sip account.x.mwi uri

Description: Sets the MWI URI that will be used for MWI subscription. If this setting is

left blank, the VSP861 uses the account x user ID for MWI subscription.

Values: SIP URI text string Default: Blank

Setting: sip account.x.network conference enable

Description: Enables or disables network conferencing for account x.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: sip_account.x.network_bridge_uri

Description: Sets the URI for the network conferencing bridge on account x.

Values: Text string (SIP URI) Default: Blank



hs_settings Module: Handset Settings

The Handset Settings allow you to configure settings for the cordless accessories that are registered to the deskset. For more information on registering cordless accessories, see the VSP861 User Guide.

General configuration file settings

Setting: hs_settings.x.handset_us_pin_code

Description: Sets the new 4-digit PIN for handset registration/deregistration.

Values: 4-digit number Default: 1592

Setting: hs settings.x.headset us pin code

Description: Sets the new 4-digit PIN for headset registration/deregistration.

Values: 4-digit number Default: 1590

MAC-specific configuration file settings

Setting: hs_settings.x.handset_name

Description: Sets the name for the handset. You can use up to 11 letters and/or

numbers. Use alphanumeric characters only—no symbol characters are

allowed.

Values: Text string Default: HANDSET



network Module: Network Settings

The network settings follow the format: network.[element].

General configuration file settings

Setting: network.vlan.wan.enable

Description: Enables or disables the WAN VLAN.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: network.vlan.wan.id

Description: Sets the WAN VLAN ID.

Values: 0–4095 **Default:** 0

Setting: network.vlan.wan.priority

Description: Sets the WAN port priority.

Values: 0–7 Default: 0

Setting: network.vlan.pc.enable

Description: Enables or disables the PC port VLAN.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: network.vlan.pc.id

Description: Sets the PC port VLAN ID.

Values: 0–4095 **Default:** 0

Setting: network.vlan.pc.priority

Description: Sets the PC port priority.

Values: 0–7 Default: 0

Setting: network.lldp med.enable

Description: Enables or disables LLDP-MED.



Setting: network.lldp med.interval

Description: Sets the LLDP-MED packet interval (in seconds).

Values: 1–30 **Default:** 10

Setting: network.eapol.enable

Description: Enables or disables 802.1x EAPOL.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: network.eapol.identity

Description: Sets the 802.1x EAPOL identity.

Values: Text string Default: Blank

Setting: network.eapol.access password

Description: Sets the 802.1x EAPOL MD5 password.

Values: Text string Default: Blank

Setting: network.vendor class id

Description: Sets the vendor ID for DHCP option 60.

Values: Text string Default: Vtech Vesa VSP861

Setting: network.user class

Description: Sets the user class for DHCP option 77.

Values: Text string Default: Vtech Vesa VSP861

Setting: network.pc port.enable

Description: Enable or disable the PC port to operate in hub/switch mode (depending

on the Enable PC Port Mirroring setting).



Setting: network.pc port.mirroring.enable

Description: When the PC port is enabled, enable PC Port Mirroring to set the port to

operate in hub mode (network traffic on the WAN port is reflected in the PC port). When Port Mirroring is not enabled, the port operates in switch

mode.

Values: 0 (disabled), 1 (enabled) Default: 0

MAC-specific configuration file settings

Setting: network.ip.mode

Setting:

Description: Sets the IPv4 network mode.

Values: disable, dhcp, static, pppoe Default: dhcp

ing: network.ip.static ip addr

Description: Sets a static IP address for the network.

Values: Text string (IPv4) Default: Blank

Setting: network.ip.subnet mask

Description: Sets the subnet mask for the network.

Values: Text string (IPv4) Default: Blank

Setting: network.ip.gateway_addr

Description: Sets the Gateway IP address.

Values: Text string (IPv4) Default: Blank

Setting: network.ip.dns1

Description: Sets the primary DNS server IP address.

Values: Text string (IPv4) Default: Blank

Setting: network.ip.dns2

Description: Sets the secondary DNS server IP address.



Setting: network.ip.manually configure dns

Description: Enable or disable manual DNS configuration.

Values: 0 (disable), 1 (enable) Default: 0

Setting: network.ip.pppoe.service_name

Description: If IPv4 mode is PPPoE, enter the name of the applicable PPPoE

provider, in case more than one is available.

Values: Text string Default: Blank

Setting: network.ip.pppoe.username

Description: If IPv4 mode is PPPoE, enter your PPPoE account username.

Values: Text string Default: Blank

Setting: network.ip.pppoe.access password

Description: If IPv4 mode is PPPoE, enter your PPPoE account password.

Values: Text string Default: Blank

Setting: network.ipv6.mode

Description: Set the IPv6 network mode, depending on how the device will be

assigned an IP address.

Values: disable, auto, static Default: disable

Setting: network.ipv6.prefix

Description: When IPv6 mode is static, enter the IPv6 address prefix length.

Values: 0–128 **Default:** 64

Setting: network.ipv6.gateway addr

Description: When IPv6 mode is static, enter the default gateway address.



Setting: network.ipv6.dns1

Description: If manual DNS configuration is enabled, enter the address for the primary

DNS server.

Values: Text string (IPv6) Default: Blank

Setting: network.ipv6.dns2

Description: If manual DNS configuration is enabled, enter the address for the

secondary DNS server.

Values: Text string (IPv6) Default: Blank

Setting: network.ipv6.manually_configure_dns

Description: Enable or disable manual DNS configuration for IPv6.

Values: 0 (disable), 1 (enable) Default: 0

Setting: network.vpn.enable

Description: If manual DNS configuration is enabled, enter the address for the

secondary DNS server.



provisioning Module: Provisioning Settings

The provisioning settings follow the format: provisioning.[element].

All these settings are exported when you manually export the configuration from the VSP861.

All the provisioning settings are included in the general configuration file.

Setting: provisioning.click to dial

Description: Enables or disables "click to dial" functionality for directory entries on the

WebUI.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: provisioning.firmware url

Description: Sets the URL for the server hosting the firmware file.

Values: Text string Default: Blank

Setting: provisioning.handset firmware url

Description: Sets the URL for the server hosting the handset firmware file.

Values: Text string Default: Blank

Setting: provisioning.fw server username

Description: Sets the authentication name for the server hosting the firmware file.

Values: Text string Default: Blank

Setting: provisioning.fw server access password

Description: Sets the authentication password for the server hosting the firmware file.

Values: Text string Default: Blank

Setting: provisioning.server_address

Description: Sets the provisioning server IP address.

Values: Text string Default: https://et.vtechphones.com/rg2



Setting: provisioning.server username

Description: Sets the authentication name for the provisioning server.

Values: Text string Default: Blank

Setting: provisioning.server_access_password

Description: Sets the authentication password for the provisioning server.

Values: Text string Default: Blank

Setting: provisioning.dhcp_option_enable

Description: Enables or disables using DHCP options for locating the configuration

and firmware files.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: provisioning.dhcp option priority 1

Description: Sets the first priority DHCP option for the provisioning/firmware file check.

Values: 66, 159, 160 **Default:** 66

Setting: provisioning.dhcp option priority 2

Description: Sets the second priority DHCP option for the provisioning/firmware file

check.

Values: 66, 159, 160 **Default:** 159

Setting: provisioning.dhcp_option_priority_3

Description: Sets the third priority DHCP option for the provisioning/firmware file check.

Values: 66, 159, 160 **Default:** 160

Setting: provisioning.resync_mode

Description: Sets the mode of the device's provisioning/firmware file check. This

determines which files the device retrieves when the resync process

begins.

Values: config_only, firmware_only, Default: config_and_firmware

config_and_firmware



Setting: provisioning.bootup check enable

Description: Enables or disables bootup check for configuration and firmware files.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: provisioning.schedule mode

Description: Sets the type of schedule check for configuration and firmware files.

Values: disable, interval, weekday Default: disable

Setting: provisioning.resync time

Description: Sets the interval (in minutes) between checks for new firmware and/or

configuration files.

Values: 0–65535 **Default:** 0 (OFF)

Setting: provisioning.weekdays

Description: Sets the day(s) when the device checks for new firmware and/or

configuration files. Enter a comma-delimited list of weekdays from 0 (Sunday) to 6 (Saturday). For example, 5,6,0 means the provisioning

check will be performed on Friday, Saturday and Sunday.

Values: 0–6 Default: Blank

Setting: provisioning.weekdays_start_hr

Description: Sets the hour when the device checks for new firmware and/or

configuration files.

Values: 0–23 **Default:** 0

Setting: provisioning.weekdays_end_hr

Description: Sets the hour when the device stops checking for new firmware and/or

configuration files.

Values: 0–23 **Default:** 0



Setting: provisioning.remote check sync enable

Description: Enables or disables remotely triggering the device to check for new firmware

and/or configuration files. The file checking is triggered remotely via a SIP

Notify message from the server containing the **check-sync** event.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: provisioning.crypto enable

Description: Enables or disables encryption check for the configuration file(s). Enable

if you have encrypted the configuration file(s) using AES encryption.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: provisioning.crypto_passphrase

Description: Sets the AES encryption passphrase for decrypting the configuration

file(s). Enter the key that was generated when you encrypted the file.

Values: Text string Default: Blank

Setting: provisioning.check trusted certificate

Description: Enables or disables accepting only a trusted TLS certificate for access to

the provisioning server.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: provisioning.pnp enable

Description: Enables or disables the VSP861 checking for the provisioning URL using

the Plug-and-Play Subscribe and Notify protocol.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: provisioning.pnp response timeout

Description: Sets how long the VSP861 repeats the SUBSCRIBE request if there is

no reply from the PnP server.

Values: 1–60 **Default:** 10



Setting: provisioning.pwd_export_enable

Description: Enables or disables passwords from being exported in plain text. This

parameter is not available on the WebUI. The passwords affected are:

network.eapol.access_password

network.ip.pppoe.access_password

tr069.acs.access_password

tr069.connection_request.access_password

provisioning.fw_server_access_password

provisioning.server_access_password

profile.admin.access_password

profile.user.access_password

sip_account.x.authentication_access_password

remoteDir.ldap_access_password

remoteDir.broadsoft_access_password

Values: 0 (disabled), 1 (enabled) Default: 0

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security Module: Security Settings

The Security settings enable you to hide PIN numbers during call operation. There are two different types of PIN masking available:

- 1. Hide DTMF digits: When Hide DTMF is enabled, PIN numbers entered during an Active Call are automatically masked out.
- 2. Password Dial: When Password Dial is enabled, PIN numbers entered as part of the dial string are hidden. For example, when entering numbers in Pre-Dial or Dial mode, PIN numbers are automatically masked out.

All the security settings are included in the general configuration file.

| Setting: | security.hide_dtmf_enable | | |
|--------------|---|----------|-------|
| Description: | Enable or disable masking all DTMF digits entered during an Active Call. | | |
| Values: | 0 (disabled), 1 (enabled) | Default: | 0 |
| Setting: | security.hide_dtmf_delay_enable | | |
| Description: | Enable to add a one-second delay before DTMF digits are masked. By default, the digits are masked immediately. | | |
| Values: | 0 (disabled), 1 (enabled) | Default: | 0 |
| Setting: | security.pwd_dial_enable | | |
| Description: | Enable to hide PIN numbers entered as part of the dial string. | | |
| Values: | 0 (disabled), 1 (enabled) | Default: | 0 |
| Setting: | security.pwd_dial_delay_enable | | |
| Description: | Enable to add a one-second delay before PIN numbers are masked. By default, the PIN number is masked immediately. | | |
| Values: | 0 (disabled), 1 (enabled) | Default: | 0 |
| Setting: | security.pwd_dial_prefix | | |
| Description: | Enter the prefix that serves as an indicator that the next x digits are masked (x being equal to the Password Dial Length). | | |
| Values: | Text string | Default: | Blank |



Setting: security.pwd_dial_length

Description: Enter the PIN number length. All digits within this length are masked. Any

digits beyond this length are not masked.

Values: 1–32 **Default:** 1



time_date Module: Time and Date Settings

The time and date settings follow the format: time_date.[element].

All these settings are exported when you manually export the configuration from the VSP861.

All the time and date settings are included in the general configuration file.

Setting: time_date.date_format

Description: Sets the format for displaying the date.

Values: DD/MM/YY, MM/DD/YY, Default: DD/MM/YY

YY/MM/DD

Setting: time date.24hr clock

Description: Enables or disables 24-hour clock.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: time_date.ntp_server

Description: Enables or disables NTP server to set time and date.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: time date.ntp server addr

Description: Sets the URL for the NTP server.

Values: Text string Default: us.pool.ntp.org

Setting: time date.ntp dhcp option

Description: Enables or disables DHCP option 42 to find the NTP server.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: time_date.selected_timezone

Description: Sets the local timezone.

Default:

America/New_York



Values: Pacific/Pago_Pago, Pacific/Honolulu,

America/Adak, America/Anchorage,
America/Vancouver, America/Tijuana,
America/Los_Angeles, America/Edmonton,
America/Chihuahua, America/Denver,
America/Phoenix, America/Winnipeg,
Pacific/Easter, America/Mexico_City,
America/Chicago, America/Nassau,
America/Montreal, America/Grand_Turk,
America/Havana, America/New_York,
America/Caracas, America/Halifax,
America/Santiago, America/Asuncion,

Atlantic/Bermuda, Atlantic/Stanley, America/Port_of_Spain, America/St_Johns,

America/Godthab,

America/Argentina/Buenos_Aires,

America/Fortaleza, America/Sao_Paulo,

America/Noronha, Atlantic/Azores, GMT,

America/Danmarkshavn, Atlantic/Faroe,

Europe/Dublin, Europe/Lisbon,

Atlantic/Canary, Europe/London,

Africa/Casablanca, Europe/Tirane,

Europe/Vienna, Europe/Brussels,

Europe/Zagreb, Europe/Prague,

Europe/Copenhagen, Europe/Paris,

Europe/Berlin, Europe/Budapest,

Europe/Rome, Europe/Luxembourg,

Europe/Skopje, Europe/Amsterdam,

Africa/Windhoek, Europe/Tallinn,

Europe/Helsinki, Asia/Gaza, Europe/Athens,

Asia/Jerusalem, Asia/Amman, Europe/Riga,

Asia/Beirut, Europe/Chisinau,

Europe/Kaliningrad, Europe/Bucharest,

Asia/Damascus, Europe/Istanbul,

Europe/Kiev, Africa/Djibouti, Asia/Baghdad,

Europe/Moscow, Asia/Tehran, Asia/Yerevan,

Asia/Baku, Asia/Tbilisi, Asia/Aqtau,

Europe/Samara, Asia/Aqtobe, Asia/Bishkek,

Asia/Karachi, Asia/Yekaterinburg,

Asia/Kolkata, Asia/Almaty, Asia/Novosibirsk,

Asia/Krasnoyarsk, Asia/Bangkok,

Asia/Shanghai, Asia/Singapore,

Australia/Perth, Asia/Seoul, Asia/Tokyo,

Australia/Adelaide, Australia/Darwin,

Australia/Sydney, Australia/Brisbane,

Australia/Hobart, Asia/Vladivostok,

Australia/Lord_Howe, Pacific/Noumea,

Pacific/Auckland, Pacific/Chatham,

Pacific/Tongatapu



Setting: time date.daylight saving auto adjust

Description: Sets the device to automatically adjust clock for daylight savings.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: time_date.daylight_saving_user_defined

Description: Enables or disables manual daylight savings configuration.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: time date.daylight saving start month

Description: Sets the month that daylight savings time starts.

Values: January-December Default: March

Setting: time date.daylight saving start week

Description: Sets the week that daylight savings time starts.

Values: 1–5 Default: 2

Setting: time date.daylight saving start day

Description: Sets the day that daylight savings time starts.

Values: Sunday, Monday, Tuesday, Default: Sunday

Wednesday, Thursday,

Friday, Saturday

Setting: time_date.daylight_saving_start_hour

Description: Sets the hour that daylight savings time starts.

Values: 00:00–23:00 **Default:** 02:00

Setting: time date.daylight saving end month

Description: Sets the month that daylight savings time ends.

Values: January–December Default: November



Setting: time date.daylight saving end week

Description: Sets the week that daylight savings time ends.

Values: 1–5 Default: 1

Setting: time_date.daylight_saving_end_day

Description: Sets the day that daylight savings time ends.

Values: Sunday, Monday, Tuesday, Default: Sunday

Wednesday, Thursday,

Friday, Saturday

Setting: time_date.daylight_saving_end_hour

Description: Sets the hour that daylight savings time ends.

Values: 00:00–23:00 **Default:** 02:00

Setting: time date.daylight saving amount

Description: Sets the daylight savings time offset in minutes.

Values: 0–255 **Default:** 60

Setting: time_date.timezone_dhcp_option

Description: Enables or disables DHCP option 2/100/101 for determining time zone

information.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: time_date.ntp_server_update_interval

Description: Sets the delay between NTP server updates, in seconds.

Values: 0–4294967295 **Default:** 1000

Setting: time date.time and date

Description: Manually sets the date and time. Use the format

<year>-<month>-<day>T<hour>:<minute>:<second>

Values: <year>-<month>-<day>T **Default:** 2016-03-01T12:00:00

<hour>:<minute>:<second>



log Module: System Log Settings

The log settings control system logging activities. System logging may be required for troubleshooting purposes. The following logging modes are supported:

- Serial/Console—system log output to an external console using a serial/RS-232 cable
- Syslog server—output to a log file on a separate server
- Volatile file

The log settings follow the format: log.[element].

All the log settings are included in the general configuration file.

Setting: log.syslog enable

Description: Enables or disables log output to syslog server.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: log.syslog_server_address

Description: Sets the syslog server IP address.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: log.syslog_server_port

Description: Sets the syslog server port.

Values: 1–65535 **Default:** 514

Setting: log.syslog_level

Description: Sets the log level. The higher the level, the larger the debug output.

5—all
4—debug
3—info
2—warning
1—error
0—critical

Values: 0–5 Default: 2



remoteDir Module: Remote Directory Settings

The remote directory settings follow the format: remoteDir.[element].

All these settings are exported when you manually export the configuration from the VSP861.

All the remote directory settings are included in the general configuration file.

Setting: remoteDir.ldap enable

Description: Enables or disables the VSP861 deskset's access to the LDAP directory.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.ldap_directory_name

Description: Sets the LDAP directory name.

Values: Text string Default: Blank

Setting: remoteDir.ldap server address

Description: Sets the LDAP server IP address.

Values: Text string Default: Blank

Setting: remoteDir.ldap port

Description: Sets the LDAP server port.

Values: 1–65535 **Default:** 389

Setting: remoteDir.ldap_protocol_version

Description: Sets the LDAP protocol version.

Values: version_2, version_3 Default: version_3

Setting: remoteDir.ldap_authentication_type

Description: Sets the LDAP authentication type.

Values: simple, ssl Default: simple



Setting: remoteDir.ldap_user_name

Description: Sets the LDAP authentication user name.

Values: Text string Default: Blank

Setting: remoteDir.ldap_access_password

Description: Sets the LDAP authentication password.

Values: Text string Default: Blank

Setting: remoteDir.ldap base

Description: Sets the LDAP search base. This sets where the search begins in the

directory tree structure. Enter one or more attribute definitions, separated by commas (no spaces). Your directory may include attributes like "cn"

(common name) or "ou" (organizational unit) or "dc" (domain component). For example, ou=accounting,dc=vtech,dc=com

Values: Text string Default: Blank

Setting: remoteDir.ldap max hits

Description: Sets the maximum number of entries returned for an LDAP search.

Limiting the number of hits can conserve network bandwidth.

Values: 0–32000 **Default:** 200

Setting: remoteDir.ldap_search_delay

Description: Sets the LDAP maximum search delay in seconds.

Values: 0–500 **Default:** 0

Setting: remoteDir.ldap_firstname_filter

Description: Sets the LDAP first name attribute filter.

Values: Text string Default: Firstname

Setting: remoteDir.ldap_lastname_filter

Description: Sets the LDAP last name attribute filter.

Values: Text string Default: Lastname



Setting: remoteDir.ldap number filter

Description: Sets the LDAP number filter.

Values: Text string Default: Blank

Setting: remoteDir.ldap_firstname_attribute

Description: Sets the name attributes. Enter the name attributes that you want the

VSP861 to display for each entry returned after an LDAP search. Separate each attribute with a space. For example, givenName sn will

display the first name and surname for each entry.

Values: Text string Default: Blank

Setting: remoteDir.ldap lastname attribute

Description: Sets the last name attributes.

Values: Text string Default: Blank

Setting: remoteDir.ldap work number attributes

Description: Sets the number attributes. Enter the number attributes that you want the

VSP861 to display for each entry returned after an LDAP search. Separate each attribute with a space. For example, "telephoneNumber mobile" will display the work phone number and mobile phone number for each entry.

Values: Text string Default: Blank

Setting: remoteDir.ldap_mobile_number_attributes

Description: Sets the mobile number attributes.

Values: Text string Default: Blank

Setting: remoteDir.ldap_other_number_attributes

Description: Sets the "other" number attributes.



Setting: remoteDir.ldap incall lookup enable

Description: Enables or disables LDAP incoming call lookup. If enabled, the VSP861

searches the LDAP directory for the incoming call number. If the number

is found, the VSP861 uses the LDAP entry for CID info.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.ldap outcall lookup enable

Description: Enables or disables LDAP outgoing call lookup. If enabled, numbers

entered in pre-dial or live dial are matched against LDAP entries. If a

match is found, the LDAP entry is displayed for dialing.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.broadsoft_enable

Description: Enables or disables the Broadsoft phonebook.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.broadsoft display name

Description: Sets the Broadsoft Phonebook display name.

Values: Text string Default: Blank

Setting: remoteDir.broadsoft server

Description: Sets the Broadsoft Phonebook IP address.

Values: Text string Default: Blank

Setting: remoteDir.broadsoft_port

Description: Sets the Broadsoft Phonebook port.

Values: 1–65535 **Default:** 0

Setting: remoteDir.broadsoft user name

Description: Sets the Broadsoft Phonebook authentication user name.



Setting: remoteDir.broadsoft access password

Description: Sets the Broadsoft Phonebook authentication password.

Values: Text string Default: Blank

Setting: remoteDir.broadsoft_dir_type

Description: Sets the Broadsoft Phonebook directory type.

Values: Group, GroupCommon, Default: Group

Enterprise,

EnterpriseCommon,

Personal

Setting: remoteDir.ldap check certificate

Description: Enables or disables accepting only a trusted LDAP certificate.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.broadsoft_check_certificate

Description: Enables or disables accepting only a trusted Broadsoft certificate.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: remoteDir.xml.x.name

Description: Sets the name of the directory as it will appear on the phone's Directory

list. For this and following parameters, x is the number of the XML

directory (1-3).

Values: Text string Default: Blank

Setting: remoteDir.xml.x.uri

Description: The location of the XML directory file, from which the phone will sync and

retrieve directory entries.

Values: URI Default: Blank

Setting: remoteDir.xml.x.call_lookup_enable

Description: Enables/disables the call lookup feature for incoming and outgoing calls.



Setting: remoteDir.xml.x.contact entry tag

Description: Sets the tag name for directory entry.

Values: Text string Default: DIR_ENTRY

Setting: remoteDir.xml.x.first_name_tag

Description: Sets the first name tag for a directory entry.

Values: Text string Default: DIR_ENTRY_NAME_FIRST

Setting: remoteDir.xml.x.last name tag

Description: Sets the last name tag for a directory entry.

Values: Text string Default: DIR_ENTRY_NAME_LAST

Setting: remoteDir.xml.x.work number tag

Description: Sets the work number tag for a directory entry.

Values: Text string Default: DIR_ENTRY_NUMBER_WORK

Setting: remoteDir.xml.x.mobile number tag

Description: Sets the mobile number tag for a directory entry.

Values: Text string Default: DIR_ENTRY_NUMBER_MOBILE

Setting: remoteDir.xml.x.other number tag

Description: Sets the other number tag for a directory entry.

Values: Text string Default: DIR_ENTRY_NUMBER_OTHER



web Module: Web Settings

The web settings control the web server IP, port, and security settings.

The web settings follow the format: web.[element].

All the web settings are included in the general configuration file.

Setting: web.server_enable

Description: Enables or disables the availability of the phone's embedded WebUI.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: web.http port

Description: Sets the http port when http is enabled.

Values: 1–65535 **Default:** 80

Setting: web.https_enable

Description: Sets server to use the https protocol.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: web.https port

Description: Sets the https port when https is enabled.

Values: 1–65535 **Default:** 443

Setting: web.activeuri_enable

Description: Enables the Action URI feature, enabling remote activation of keys and

features when a server sends an HTTP GET.



trusted_ip Module: Trusted IP Settings

The trusted_ip settings provide enhanced security for the VSP861. When enabled, these settings can filter network traffic and reject any traffic from unauthorized sources.

The trusted_ip settings follow the format: trusted_ip.[element].

All the trusted_ip settings are included in the general configuration file.

Setting: trusted ip.only accept allowed ip

Description: Enables or disables using the Allowed IP list to filter network traffic. When

enabled, all unsolicited IP traffic will be blocked unless it is from one of

the trusted IP addresses on the "Allowed IP" list.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: trusted_ip.x.allow_ip

Description: Enter an IP address or address range for one instance of the "Allowed IP"

list. x ranges from 1 to 10. See "Trusted IP" on page 126 for more

information.

Values: Text string (IPv4 or IPv6, IP **Default**: Blank

range in IPv4 or IPv6)



trusted_servers Module: Trusted Server Settings

The trusted_servers setting provides enhanced security for the VSP861. When enabled, this setting can filter network traffic and reject any traffic from unauthorized sources.

The trusted servers setting follows the format: trusted servers.[element].

The trusted_servers setting is included in the general configuration file.

Setting: trusted_servers.only_accept_sip_account_servers

Description: Enables or disables using each enabled account's Registration server,

SIP server, Outbound Proxy server and Backup Outbound Proxy server

as sources for trusted SIP traffic.



user_pref Module: User Preference Settings

The user settings are accessible to the VSP861 user. These settings are useful for initial setup. You may wish to remove these settings from auto-provisioning update files so that users do not have their own settings overwritten.

The user preference settings follow the format: user_pref.[element].

The user preference settings (except for user_pref.call_terminated.busy_tone_enable) are exported when you manually export the configuration from the VSP861.

General configuration file settings

Setting: user pref.account.x.ringer

Description: Sets the ring tone for account x.

Values: 1–10 **Default:** 1

Setting: user_pref.web_language

Description: Sets the language that appears on the WebUI.

Values: en, en-GB, es-MX, es, fr-CA, **Default:** en

fr, de, it, pt, nl, el, ru, tr, pl

Setting: user pref.language

Description: Sets the language that appears on the device screen.

Values: en, en-GB, es-MX, es, fr-CA, **Default:** en

fr, de, it, pt, nl, el, ru, tr, pl

Setting: user pref.notify.led.missed call.enable

Description: Sets how the Message Waiting LED operates. When enabled, the LED

turns on for missed calls and new messages. When disabled, the LED

turns on for new messages only.

Note: This setting is not available on the phone menu or WebUI.



Setting: user pref.text input option

Description: Sets the order and available language input options available when users

edit or enter text on the LCD.

Note: This setting is not available on the phone menu or WebUI, and applies to models sold and installed outside North America only.

Values: number,uc_western,

Default:

number,lc_western,

uc_western

lc_western,uc_ru,lc_ru,

uc_el,lc_el

Setting: user pref.call terminated.busy tone enable

Description: Enables the VSP861 to play a busy tone when the far-end party ends the

call, or when a network error condition (keep-alive failure) occurs.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: user pref.account.x.diversion display

Description: Enables or disables the display of diversion <name-addr> info (if

available) for calls forwarded to account x.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: user pref.blf indication option

Description: Configures the BLF LED behavior for different service providers. Two

options are available, as described below.

Option 1 behavior

Registration Error: Blinking ORANGE

(every 2 secs)

Incoming call: Fast ORANGE flash
Held call: Fast ORANGE flash*
Active call: Steady ORANGE

Idle: Off

* behavior applies only to certain servers.

Option 2 behavior

Registration Error: Off

Incoming call: Fast ORANGE flash Held call: Fast ORANGE flash* Active call: Steady ORANGE

Idle: Steady GREEN

Values: 1, 2 Default:

Setting: user pref.quick transfer

Description: Sets transfer options for Quick Dial and BLF Programmable keys during an

active call.

Values: new call, blind, attended **Default**: new call



MAC-specific configuration file settings

Setting: user pref.backlight timeout

Description: Sets the backlight timeout in seconds.

Values: 10–60 **Default:** 30

Setting: user pref.audio mode

Description: Sets the default audio mode.

Values: speaker, headset Default: speaker

Setting: user pref.hold reminder.enable

Description: Enables or disables audible hold reminder.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: user pref.hold reminder.interval

Description: Sets the interval for the audible hold reminder in seconds.

Values: 10–300 **Default:** 30

Setting: user pref.call waiting.tone enable

Description: Enables or disables the call waiting tone.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: user_pref.call_waiting.tone_interval

Description: Sets the interval for the call waiting tone in seconds.

Values: 10–60 **Default:** 30

Setting: user pref.call waiting.mode

Description: Enables or disables rejecting calls if already on a call.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: user pref.lcd contrast

Description: Sets the LCD contrast on the VSP861.

Values: 1–7 Default: 4



Setting: user_pref.backlight

Description: Sets the backlight brightness level.

Values: off, low, medium, high Default: high

Setting: user_pref.idle_backlight

Description: Sets the backlight brightness level when the VSP861 is idle.

Values: off, low, medium, high Default: off

Setting: user pref.absent timeout

Description: Sets the absent timeout (the interval after going off hook with no action

taken) in seconds. After the absent timeout, the phone returns to idle mode.

Values: 10–60 **Default:** 30

Setting: user pref.speaker volume

Description: Sets the speakerphone volume.

Values: 1–9 Default: 5

Setting: user pref.headset volume

Description: Sets the headset volume.

Values: 1–9 Default: 5

Setting: user pref.handset volume

Description: Sets the corded handset volume.

Values: 1–9 Default: 5

Setting: user_pref.key_beep_enable

Description: Enables or disables key beeps on the VSP861.



call_settings Module: Call Settings

The call settings configure data related to a user's call preferences. The data is stored internally at /mnt/flash/CallSettings.xml.

All the call settings (except one) follow the format: call_settings.account.x.[element] where x is an account number ranging from 1 to .

General configuration file settings

Setting: call_settings.account.x.call_completion_enable

Description: Enables or disables the Call Completion for Busy Subscribers (CCBS)

feature. The feature notifies users when a previously busy number is available and asks them whether they wish to call the number back.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call_settings.account.x.call_completion_alert_enable

Description: Enables or disables an audible alert (similar to a hold reminder alert tone)

if the user is on another call when the auto redial interval expires.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: call settings.account.x.auto redial interval

Description: Sets the countdown timer (in seconds) until the user is prompted for the

next dialing attempt.

Values: 1–300 **Default:** 30

Setting: call settings.account.x.auto redial repeat

Description: Sets how many auto redial attempts are made.

Values: 1–30 **Default:** 10

MAC-specific configuration file settings

Setting: call settings.account.x.block anonymous enable

Description: Enables or disables anonymous call blocking.



Setting: call settings.account.x.outgoing anonymous enable

Description: Enables or disables outgoing anonymous calls.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call settings.account.x.dnd enable

Description: Enables or disables Do Not Disturb for account x.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call settings.account.x.dnd incoming calls

Description: Sets whether incoming calls are shown or rejected when DND is on for

account x.

Values: show, reject Default: reject

Setting: call settings.account.x.call fwd always enable

Description: Enables or disables Call Forward Always for account x.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call settings.account.x.call fwd always target

Description: Sets the Call Forward Always target number for account x.

Values: Text string Default: Blank

Setting: call settings.account.x.call fwd busy enable

Description: Enables or disables Call Forward Busy for account x.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call_settings.account.x.call_fwd_busy_target

Description: Sets the Call Forward Busy target number for account x.

Values: Text string Default: Blank

Setting: call_settings.account.x.cfna_enable

Description: Enables or disables Call Forward No Answer for account x.



Setting: call settings.account.x.cfna target

Description: Sets the Call Forward No Answer target number for account x.

Values: Text string Default: Blank

Setting: call_settings.account.x.cfna_delay

Description: Sets the Call Forward No Answer delay (in number of rings) for account x.

Values: 1–10 Default: 6

Setting: call settings.missed call alert enable

Description: Enables or disables missed call alerts.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: call settings.hotline enable

Description: Enables or disables the hotline feature.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call settings.hotline account

Description: Sets the account used for dialing the hotline number.

Values: 0- Default: 0 (default account)

Setting: call settings.hotline number

Description: Sets the number dialed by the hotline feature.

Values: Text string Default: Blank

Setting: call settings.hotline delay

Description: Sets the delay (in seconds) between the phone going off hook and the

hotline number being dialed.

Values: 0–10 Default: 0



Setting: call settings.account.x.unconditional auto answer enable

Description: Enables or disables unconditional Auto Answer. Auto Answer allows a

deskset or conference phone to automatically answer incoming calls to that account without user intervention. An auto answer tone will sound.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call settings.account.x.unconditional auto answer delay

Description: Sets the delay before the phone auto answers a call.

Values: 0–30 Default: 2

Setting: call settings.account.x.unconditional auto answer mute on ans

Description: Enables or disables muting the mic upon auto answering.

Feature Keys" on page 15.



pfk Module: Programmable Feature Key Settings

The programmable feature key (PFK) settings store the data associated with each programmable key. On ErisTerminal desksets, you can program the programmable feature keys.

All the programmable feature key settings are included in the general configuration file.

Programmable Feature Keys

The programmable feature key settings follow the format: pfk.x.[element], where x is the programmable feature key ID, ranging from 1 to 30.

Setting: pfk.x.feature

Description: Assigns a feature to PFK x.

Values: unassigned, line, dir, call Default: See "Programmable"

log, redial, messages, dnd, dnd all, cfwd all, cfwd busy, cfwd no answer, quick dial, busy lamp field, acd, page, multicast page, park call, retrieve parked call, in call dtmf, callback, group call pickup, direct call pickup,

prefix_dial, chp, hg, sf, lock_key, flash, xml app

Setting: pfk.x.quick dial

Description: Sets the quick dial string to use if quick dial is assigned to PFK x.

Values: Text string (SIP URI) Default: Blank

Setting: pfk.x.blf

Description: Sets the BLF string to use if Busy Lamp Field is assigned to PFK x.

Values: Text string (SIP URI) Default: Blank

Setting: pfk.x.incall dtmf

Description: Sets the DTMF string if In-call DTMF is assigned to PFK x.

Values: Text string (SIP URI) Default: Blank



Setting: pfk.x.multicast_zone

Description: Sets the multicast paging zone if multicast page is assigned to PFK x.

Values: 1–10 Default: Blank

Setting: pfk.x.account

Description: Sets the SIP account used for the assigned feature (if applicable).

Values: 1–8 Default: 1

Setting: pfk.x.page destination

Description: If required by your service provider, enter a page destination number.

The target number for the outgoing page will be formed by concatenating Paging feature access code of the selected account and the PFK page

destination value.

Values: text string Default: blank

Setting: pfk.x.park destination

Description: If required by your service provider and sip_account.x.park_variant

setting, enter a value for the park "orbit" or extension.

Values: text string Default: blank

Setting: pfk.x.park_retrieval_source

Description: If required by your service provider and sip account.x.park variant

setting, enter a value for the park "orbit" or extension.

Values: text string Default: blank

Setting: pfk.x.prefix

Description: Enter a prefix to be dialed (hidden from the user) when the key is

pressed. The prefix is added to any user-entered digits. **%N** can be used for substitution of user-entered digits. For example, ***71%N#** uses [***71**] +

[user-entered digits] + [#] as the outgoing dialing string.

Values: text string Default: blank



Setting: pfk.x.call handling profile

Description: If pfk.x.feature is chp, enter the string of the call handling profile that the

pfk LED will indicate.

Values: text string Default: blank

Setting: pfk.x.call handling profile set code

Description: If pfk.x.feature is chp, enter the call handling profile FAC and profile index

number to activate the profile.

Values: text string Default: blank

Setting: pfk.x.hunt_group

Description: If pfk.x.feature value is hg, enter the hunt group extension number

assigned for this pfk.

Values: text string Default: blank

Setting: pfk.x.secretarial filtering

Description: If pfk.x.feature value is sf, enter the manager's extension number

assigned for this pfk

Values: text string Default: blank

Setting: pfk.x.xml_uri

Description: If pfk.x.feature is xml app, enter the URI to fetch the XML application to

be executed.

Values: text string Default: blank

Setting: pfk.x.direct_pickup

Description: If pfk.x.feature is direct call pickup, enter the Direct Call Pickup feature

access code (FAC).

Values: text string Default: blank



application Module: Application Shortcuts

The application shortut settings store the data associated with each application shortcut displayed on the three pages of the Home screen.

Application Shortcuts

The application shortut settings follow the format: application.x.[element], where x is the application shortcut key ID, ranging from 1 to 30.

Setting: application.x.feature

Description: Assigns a feature to application shortcut.

Values: unassigned, new call, Default: app 1: new call

redial, dir, call center,
settings, status, messages,
dnd all, call log, cfwd all,
callback, blacklist, pfk, line,
multicast page, select line,
app 2: call center
app 3: pfk
app 4: settings
app 5: messages
app 6: dir

select zone, call recording, select default line app 7: dnd all

app 8: cfwd all app 9: call recording app 10: call log app 11: select line app 15: status

app 16: select zone app 17: blacklist app 20: redial

app 12-14, 18-19, 21-30:

unassigned

Setting: application.x.account

Description: Sets the SIP account used for the assigned feature (if applicable).

Values: 1–8 Default: 1

Setting: application.x.multicast zone

Description: Sets the multicast paging zone if multicast page is assigned to

application shortcut.

Values: 1–10 Default: Blank



speed_dial Module: Speed Dial Settings

The speed dial key settings configure the dial pad keys for speed dialing pre-programmed phone numbers. When configured, the VSP861 user can press and hold a dial pad key to dial a programmed phone number.

The speed dial key settings follow the format speed_dial.x.[element], where x is the dial pad key, ranging from 1 to 0 (with 0 being the "0" OPER key).

All the speed dial settings are included in the MAC-specific configuration file.

Setting: speed_dial.x.name

Description: Sets the name associated with the phone number for dial pad key x. The

name is visible on the VSP861 screen.

Values: Text string Default: Blank

Setting: speed dial.x.number

Description: Sets the phone number that dial pad key x dials when pressed and held.

Values: Text string (SIP URI) Default: Blank

Setting: speed dial.x.account

Description: Sets the SIP account used for dialing when dial pad key x is pressed and

held.

Values: 0- Default: 0

(0 is the default account)



audio Module: Audio Settings

The audio settings include jitter buffer parameters and RTP port settings.

All the audio settings are included in the general configuration file.

Setting: audio.x.jitter_mode

Description: Select the desired mode for the jitter buffer: fixed (static) or adaptive. This

setting depends on your network environment and conditions.

Values: fixed, adaptive Default: adaptive

Setting: audio.x.fixed_jitter.delay

Description: When in fixed jitter buffer mode, set the delay (in ms) desirable to provide

good audio quality with the minimal possible delay.

Values: 30–500 **Default:** 70

Setting: audio.x.adaptive_jitter.min_delay

Description: When in adaptive jitter buffer mode, set the minimum delay (in ms)

desirable to maintain data packet capture and audio quality.

Values: 20–250 **Default:** 60

Setting: audio.x.adaptive jitter.target delay

Description: When in adaptive jitter buffer mode, set the target delay (in ms) desirable

to provide good audio quality with the minimal possible delay.

Values: 20–500 **Default:** 80

Setting: audio.x.adaptive jitter.max delay

Description: When in adaptive jitter buffer mode, set the maximum delay (in ms)

desirable to maintain data packet capture and audio quality.

Values: 180–500 **Default:** 240

Setting: audio.x.rtp.port start

Description: Sets the Local RTP port range start.

Values: 1–65535 **Default:** 18000



Setting: audio.x.rtp.port_end

Description: Sets the Local RTP port range end.

Values: 1–65535 **Default:** 19000

Setting: audio.x.rtcp_xr.enable

Description: Enables or disables reporting of RTCP XR via SIP to a collector server.

RTP Control Protocol Extended Reports (RTCP XR) are used for voice

quality assessment and diagnostics.



ringersetting Module: Distinctive Ringer Settings

The distinctive ringer settings configure the distinctive ringer feature. For more information, see "Ringer Settings" on page 78. You can configure up to 8 instances of the distinctive ringer feature.

The ringer setting parameters follow the format ringersetting.x.[element], where x is the instance of the setting, ranging from 1 to 8.

All the ringer settings are included in the general configuration file.

Setting: ringersetting.x.ringer text

Description: Enter the text that will match the "info" parameter and play the ringer

tone. The matching of the "info" parameter and ringer_text setting is case

sensitive.

Values: Text string in the format Default: Blank

ringerx (e.g., ringer1)

Setting: ringersetting.x.ringer_type

Description: Select the desired ring tone for ringer setting x.

Values: 1–10 Default: 1



call_record Module: Call Recording Settings

The call recording settings configure call recording for the VSP861. Ensure that a microSD is also inserted into the unit.

All the call recording settings are included in the general configuration file.

Setting: call record.enable

Description: Enables call recording on the deskset.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: call record.tone enable

Description: Enables a call record tone, to be played over the line when recording

begins and ends.



file Module: Imported File Parameters

The "file" parameters enable the provisioning file to import additional configuration files of various types, including:

- Contact lists
- Custom logos
- Security certificates

Certificates can be added via provisioning. There are two types of certificate:

- Trusted: Trusted Certificates are for server authentication with secured HTTP transaction in the following applications: SIP signaling, Provisioning, Firmware, LDAP directory service, and Broadsoft directory service. Up to 20 trusted certificates can be installed.
- Device: A single Device Certificate can be uploaded so that other parties can authenticate the phone in the following cases:
 - When the phone acts as a web server for the user to manage configurations.
 - When the phone acts as a client for applications where HTTP is supported.

None of these settings are exported when you manually export the configuration from the VSP861.

General configuration file settings

| Setting: | file.certificate.x.url | | |
|--------------|--|----------|-------|
| Description: | URL to upload a trusted certificate file in pem or crt. It will be given index x and marked as unprotected. x ranges from 1 to 20. | | |
| Values: | Text string | Default: | Blank |
| | | | |
| Setting: | file.protected_certificate.x.url | | |
| Description: | URL to upload a trusted certificate file in pem or crt. It will be given index x and marked as protected. x ranges from 1 to 20. | | |
| Values: | Text string | Default: | Blank |



Setting: file.certificate.trusted.url

Description: URL to upload a trusted certificate file in pem or crt. It will be given the

first available index and marked as unprotected.

Values: Text string Default: Blank

Setting: file.protected certificate.trusted.url

Description: URL to upload a trusted certificate file in pem or crt. It will be given the

first available index and marked as protected.

Values: Text string Default: Blank

Setting: file.protected certificate.custom device.url

Description: URL to upload a custom device certificate to override the factory installed

device certificate.

Values: Text string Default: Blank

Setting: file.action

Description: Enables you to delete certain certificates.

 removecertificate_customdevice: remove the custom device certificate and resume the use of the factory installed device

 removecertificate_allnonprotected: remove all non-protected trusted certificates

 removecertificate_all: remove the custom device certificate and all protected or non-protected trusted certificates

Enables you to delete a custom language from the WebUI, the deskset screens, or both.

Values: removecertificate_ Default: Blank

customdevice, removecertificate_

allnonprotected, removecertificate_all

removecustomlanguage_all, removecustomlanguage_webui, removecustomlanguage_desksetui



Setting: file.bootup logo

Description: URL of custom logo shown during bootup. For logo specifications, see

"Logo specifications" on page 21.

To restore the default "VTech" logo, set the value to %NULL in the configuration file. For example: file.bootup logo = %NULL

Values: Text string Default: Blank

Setting: file.idle logo

Description: URL of custom logo shown on the idle screen. For logo specifications,

see "Logo specifications" on page 21.

To restore the default "VTech" logo, set the value to "NULL in the configuration file. For example: file.bootup logo = %NULL

Values: Text string Default: Blank

Setting: file.language.deskset.url

Description: URL of the Deskset UI Custom Language file to be imported.

Values: Text string Default: Blank

Setting: file.language.webui.url

Description: URL of Web UI Custom Language file to be imported.

Values: Text string Default: Blank

Setting: file.vpn.advanced config

Description: URL of OpenVPN client configuration file. For more information, see

"VPN" on page 92.

Values: Text string Default: Blank

MAC-specific configuration file settings

Setting: file.custom ringer

Description: Enter URI to WAV file for a custom ringer. The custom ringer replaces

Ringtone 10 on Ringer tone menu. For more information about file

format, see "Custom Ringer" on page 68.

To restore the default "VTech" logo, set the value to "NULL in the configuration file. For example: file.bootup logo = %NULL

Values: Text string Default: Blank



Setting: file.contact.directory.append

Description: URL of contact directory to be imported. Entries in the imported file will

be added to existing directory entries.

Values: Text string Default: Blank

Setting: file.contact.directory.overwrite

Description: URL of contact directory to be imported. Entries in the imported file will

replace all existing directory entries.

Values: Text string Default: Blank

Setting: file.contact.blacklist.append

Description: URL of contact blacklist to be imported. Entries in the imported file will be

added to existing blacklist entries.

Values: Text string Default: Blank

Setting: file.contact.blacklist.overwrite

Description: URL of contact blacklist to be imported. Entries in the imported file will

replace all existing directory entries.

Values: Text string Default: Blank



xml_app Module: XML App Settings

The VSP861 supports both push and pull server applications. The XML app settings allow you to enable "push" events and how they interact with the phone during calls.

The XML app settings are included in the general configuration file.

Setting: xml_app.http_push_enable

Description: Enable or disable HTTP push, which enables the phone to display XML

objects that are "pushed" to the phone from the server via http/https

POST or SIP NOTIFY.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: xml app.push during call enable

Description: Enable or disable the phone to display pushed XML objects during a call.

Otherwise, the XML application is displayed after the call is over.



system_event Module: Action URI Settings

You can enter Action URIs to allow the VSP861 to interact with a server application by using an HTTP GET request. The action URI triggers a GET request when a specified event occurs. Action URIs allow an external application to take control of the display when an event occurs. For more information, see "Server Application" on page 81.

All the Action URI settings are included in the general configuration file.

Setting: system_event.startup.action_uri

Description: Enter URI for GET request triggered at end of phone bootup.

Values: Text string Default: Blank

Setting: system event.registered.action uri

Description: Enter URI for GET request triggered at end of line registration.

Values: Text string Default: Blank

Setting: system event.on hook.action uri

Description: Enter URI for GET request triggered when phone goes from active to idle.

Values: Text string Default: Blank

Setting: system event.off hook.action uri

Description: Enter URI for GET request triggered when phone goes into dial mode.

Values: Text string Default: Blank

Setting: system event.incoming call.action uri

Description: Enter URI for GET request triggered for incoming calls or call waiting

events.

Values: Text string Default: Blank

Setting: system event.outgoing call.action uri

Description: Enter URI for GET request triggered when phone sends SIP INVITE

message.

Values: Text string Default: Blank



Setting: system_event.poll.action_uri

Description: Enter URI for GET request.

Values: Text string Default: Blank

Setting: system_event.poll.interval

Description: Enter interval (in seconds) between poll.action_uri requests.

Values: 1–65535 **Default:** 3600

Setting: system_event.connected.action_uri

Description: Enter URI for GET request triggered when phone has active call or is

paging.

Values: Text string Default: Blank

Setting: system event.registration event.action uri

Description: Enter URI for GET request triggered when the registration state changes.

Values: Text string Default: Blank



tr069 Module: TR-069 Settings

The Broadband Forum's Technical Report 069 (TR-069) defines a protocol for remote management and secure auto-configuration of compatible devices. The TR-069 settings allow you to enable TR-069 and configure access to an auto-configuration server (ACS).

All the TR-069 settings are included in the general configuration file.

Setting: tr069.enable

Description: Enable/disable the TR-069 subsystem.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: tr069.acs.url

Description: Enter the URL to the auto configuration server (ACS).

Values: Text string Default: Blank

Setting: tr069.acs.username

Description: Enter user name for ACS authentication.

Values: Text string Default: Blank

Setting: tr069.acs.access_password

Description: Enter password for ACS authentication.

Values: Text string Default: Blank

Setting: tr069.periodic inform.enable

Description: Enable/disable the phone sending Inform messages to the server.

Values: 0 (disabled), 1 (enabled) Default: 0

Setting: tr069.periodic inform.interval

Description: Set the interval (in seconds) between sending Inform messages.

Values: 1–65535 **Default:** 3600



Setting: tr069.connection_request.username

Description: Set the user name for authenticating the connection sent from the ACS.

Values: Text string Default: Blank

Setting: tr069.connection_request.access_password

Description: Set the password for authenticating the connection sent from the ACS.

Values: Text string Default: Blank



tone Module: Tone Definition Settings

The Tone Definition settings configure data for various tones for the purpose of localization. The Audio Manager component uses the data from this model to populate the mcu on bootup.

Each tone definition must be at least one element containing a string of 12 element attributes separated by a space:

"<num of freq> <freq1> <amp1> <freq2> <amp2> <freq3> <amp3> <freq4> <amp4>
<on duration> <off duration> <repeat count>"

Where:

```
<num of freq>: 0-2
<freq1>: 0-65535 (Hz)
<amp1>: -30-6 (dB)
<freq2>: 0-65535 (Hz)
<amp2>: -30-6 (dB)
<freq3>: 0 (for future development—modifying attribute has no effect)
<amp3>: 0 (for future development—modifying attribute has no effect)
<freq4>: 0 (for future development—modifying attribute has no effect)
<amp4>: 0 (for future development—modifying attribute has no effect)
<amp4>: 0 (for future development—modifying attribute has no effect)
<on duration>: 0-65535 (milliseconds)
<off duration>: 0-65535 (milliseconds)
<repeat count>: 0-65535
```

All the tone definition settings are included in the general configuration file.

```
Setting: tone.call_waiting_tone.num_of_elements

Description: Sets the number of elements for the call waiting tone.

Values: 1-5 Default: 1

Setting: tone.call_waiting_tone.element.1

Description: Defines the call waiting tone element 1.

Values: Tone element string Default: 1 440 -22 0 0 0 0 0 500 0 1
```



Setting: tone.call waiting tone.element.x

Description: Defines the call waiting tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.call_waiting_tone.num_of_repeat_all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.hold_reminder_tone.num_of_elements

Description: Sets the number of tone elements for the hold reminder tone.

Values: 1–5 Default: 1

Setting: tone.hold reminder tone.element.1

Description: Defines the hold reminder tone element 1.

Values: Tone element string **Default**: 1 770 -22 0 0 0 0 0 300 0 1

Setting: tone.hold reminder tone.element.x

Description: Defines the hold reminder tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.hold reminder tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.inside dial tone.num of elements

Description: Sets the number of tone elements for the secondary dial tone (see "Dial

Plan" on page 54 for description and behavior).

Values: 1–5 Default: 1



Setting: tone.inside dial tone.element.1

Description: Defines the secondary dial tone element 1.

Values: Tone element string **Default:** 2 440 -22 350 -22 0 0 0 0

65535 0 65535

Setting: tone.inside dial tone.element.x

Description: Defines the secondary dial tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.inside dial tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.stutter dial tone.num of elements

Description: Sets the number of tone elements for the stutter dial tone.

Values: 1–5 Default: 2

Setting: tone.stutter_dial_dial_tone.element.1

Description: Defines the stutter dial tone element 1.

100 10

Setting: tone.stutter_dial_dial_tone.element.2

Description: Defines the stutter dial tone element 2.

Values: Tone element string Default: 2 440 -22 350 -22 0 0 0 0

65535 0 65535

Setting: tone.stutter dial tone.element.x

Description: Defines the stutter dial tone element x (x = 3-5).

Values: Tone element string Default: Blank



Setting: tone.stutter dial tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.busy tone.num of elements

Description: Sets the number of tone elements for the busy tone.

Values: 1–5 Default: 1

Setting: tone.busy tone.element.1

Description: Defines the busy tone element 1.

Values: Tone element string Default: 2 480 -22 620 -22 0 0 0 0 500

500 65535

Setting: tone.busy tone.element.x

Description: Defines the busy tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.busy_tone.num_of_repeat_all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.ring_back_tone.num_of_elements

Description: Sets the number of tone elements for the ringback tone.

Values: 1–5 Default: 1

Setting: tone.ring back tone.element.1

Description: Defines the ringback tone element 1.

Values: Tone element string **Default:** 2 440 -22 480 -22 0 0 0 0

2000 4000 65535



Setting: tone.ring back tone.element.x

Description: Defines the ringback tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.ring_back_tone.num_of_repeat_all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.dial tone.num of elements

Description: Sets the number of tone elements for the dial tone.

Values: 1–5 **Default:** 1

Setting: tone.dial tone.element.1

Description: Defines the dial tone element 1.

Values: Tone element string **Default:** 2 440 -22 350 -22 0 0 0 0

65535 0 65535

Setting: tone.dial tone.element.x

Description: Defines the dial tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.dial_tone.num_of_repeat_all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.record tone.num of elements

Description: Sets the number of tone elements for the record tone.

Values: 1–5 Default: 1



Setting: tone.record tone.element.1

Description: Defines the record tone element 1.

Values: Record element string Default: 1 770 -22 0 0 0 0 0 800 0 1

Setting: tone.record_tone.element.x

Description: Defines the record tone element x (x = 2-5).

Values: Record element string Default: Blank

Setting: tone.record tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

Setting: tone.congestion tone.num of elements

Description: Sets the number of tone elements for the congestion tone.

Values: 1–5 Default: 3

Setting: tone.congestion tone.element.1

Description: Defines the dial tone element 1.

Values: Tone element string Default: 1 950 -22 0 0 0 0 0 330 0 1

Setting: tone.congestion tone.element.2

Description: Defines the dial tone element 2.

Values: Tone element string **Default:** 1 1400 -22 0 0 0 0 0 330 0 1

Setting: tone.congestion_tone.element.3

Description: Defines the dial tone element 3.

Values: Tone element string Default: 1 1800 -22 0 0 0 0 0 330 1000 1

Setting: tone.congestion_tone.element.x

Description: Defines the dial tone element x (x = 4-5).

Values: Tone element string Default: Blank



Setting: tone.congestion tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 65535

Setting: tone.auto answer tone.num of elements

Description: Sets the number of tone elements for the auto answer tone.

Values: 1–5 Default: 1

Setting: tone.auto answer tone.element.1

Description: Defines the dial tone element 1.

Values: Tone element string Default: 2 500 -22 800 -22 0 0 0 0 1000 0 1

Setting: tone.auto answer tone.element.x

Description: Defines the dial tone element x (x = 2-5).

Values: Tone element string Default: Blank

Setting: tone.auto answer tone.num of repeat all

Description: Sets the number of repeats of all elements in sequence; that is, repeating

back to the first element.

Values: 0–65535 **Default:** 0

profile Module: Password Settings

The password settings allow you to set the default administrator and user passwords in the configuration file. The administrator password is usually included in the general configuration file, while the user password is usually included in the MAC-specific configuration file. The passwords can also be set using the WebUI. Be aware that scheduled provisioning configuration file updates may reset these passwords.



General configuration file settings

Setting: profile.admin.access_password

Description: Sets the administrator password for accessing the admin menus on the

VSP861 and the WebUI.

Values: Text string Default: admin

(15 characters maximum)

MAC-specific configuration file settings

Setting: profile.user.access_password

Description: Sets the user password for logging on to the WebUI and editing

user-accessible settings.

Values: Text string Default: user

(15 characters maximum)



page_zone Module: Paging Zone Settings

The paging zone settings allow you to define a maximum of 10 paging zones that the VSP861 can use for multicast paging.

The paging zone parameters (except for page_zone.call_priority_threshold) follow the format page_zone.x.[element], where x is the paging zone ID number, ranging from 1 to 10.

All the paging zone settings are included in the general configuration file.

Setting: page_zone.x.name

Description: Sets the paging zone name, which appears on VSP861 LCD for outgoing

and incoming multicast pages. A maximum of 15 characters is allowed.

Values: Text string Default: Blank

Setting: page zone.x.multicast address

Description: Enter the multicast IP address that the VSP861 will monitor. The range of

valid IP addresses is 224.0.0.0 to 239.255.255.255.

Values: IPv4, IPv6 or FQDN Default: Blank

Setting: page zone.x.multicast port

Description: Enter the multicast port associated with the multicast IP. The range of

valid ports is 1 to 65535.

Values: 1–65535 **Default**: Blank

Setting: page_zone.x.accept_incoming_page

Description: Enables or disables the VSP861 from receiving incoming multicast pages

for that paging zone. If disabled, the VSP861 can make outgoing

multicast pages only.

Values: 0 (disabled), 1 (enabled) Default: 1

Setting: page zone.x.priority

Description: Set the paging zone priority from 1 to 10. Zones with a priority higher than

another zone can interrupt the lower-priority zone's active page.

Values: 1–10 Default: 5



Setting: page_zone.call_priority_threshold

Description: Set the call_priority_threshold. If the paging zone priority

(page_zone.x.priority) is higher or equal to the call priority, then a multicast page can interrupt an active, dialing, or incoming call.

Values: 1–10 Default: 2



phonelock Module: Phone Lock Settings

The Phone Lock feature restricts certain hard keys and features unless the user enters a PIN code. For more information about phone lock modes, see "Using the Phone Lock menu" on page 41.

All the phone lock settings are part of the general configuration file.

Setting: phonelock.type

Description: Sets the phone lock type.

Values: disabled, restricted_config, Default: disabled

restricted_call,

emergency_call_only

Setting: phonelock.pin

Description: Sets the pin for unlocking the phone.

Values: 4 to 15 digits Default: 1234

Setting: phonelock.autolock_timeout

Description: Sets the delay (in seconds) before the phone locks when idle.

Values: 0 (disabled)–3600 **Default:** 0

Setting: phonelock.restricted_account

Description: Sets the restricted account.

Values: 0 (default)-4 Default: 0



CHAPTER 6

TROUBLESHOOTING

If you have difficulty with your VSP861 deskset, please try the suggestions below.



For customer service or product information, contact the person who installed your system. If your installer is unavailable, visit our website at businessphones.vtech.com or call 1 (888) 370-2006.

Common Troubleshooting Procedures

Follow these procedures to resolve common issues. For more troubleshooting information, see the user's manual for your product.

Screen is blank.

Ensure power is connected. If powered by an AC adapter, check that the adapter is plugged into a wall socket and the VSP861 power jack. If powered by PoE, ensure that the network switch is providing power through the correct ports.

The DECT headsetdoesn't register. "Registration failed" appears on the screen.

- Ensure the headset is fully charged and in the charger. Remove and replace the headset in its charger before selecting **Register** on the VSP861.
- Ensure the headset is not already registered to another phone. If it has been registered to another phone, deregister it.

The DECT handset doesn't register. "Registration failed" appears on the screen.

- Ensure the handset is fully charged and in the charger.
- Ensure the handset is not already registered to another base. If it has been registered to another base, deregister it.

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Pages are not received.

The Page auto answer setting is set to Manual. Check the General Account Settings.

Calls are answered on the headset rather than the speakerphone after I press a Line key or Accept to answer a call.

Change the audio mode from Headset to Speaker. On the VSP861 Home screen,
 press Settings > User Settings > Audio > Audio mode.

My computer can't connect to the network after plugging the Ethernet cable through the PC port.

- Make sure the VSP861 is connected to power. The PC port does not work when the VSP861 does not have power source or during a power outage.
- Make sure you plug the Ethernet cable connected to the router into the VSP861
 Ethernet port and the Ethernet cable connected to the computer into the VSP861
 PC port.

The firmware upgrade or configuration update isn't working.

- Before using the WebUI, ensure you have the latest version of your web browser installed. Some menus and controls in older browsers may operate differently than described in this manual.
- Ensure you have specified the correct path to the firmware and configuration files on the SERVICING > Firmware Upgrade > Auto Upgrade page and the SERVICING > Provisioning page.
- If the phone is not downloading a MAC-specific configuration file, ensure the filename is all upper case.

Provisioning: "Use DHCP Option" is enabled, but the VSP861 is not getting a provisioning URL from the DHCP Server.

Ensure that DHCP is enabled in Network settings.

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APPENDIXES

Appendix A:Maintenance

Taking care of your telephone

- Your VSP861 deskset contains sophisticated electronic parts, so you must treat it with care.
- Avoid rough treatment.
- Place the corded handset down gently.
- Save the original packing materials to protect your VSP861 deskset if you ever need to ship it.

Avoid water

You can damage your VSP861 deskset if it gets wet. Do not use the corded handset in the rain, or handle it with wet hands. Do not install the VSP861 deskset near a sink, bathtub or shower.

Electrical storms

■ Electrical storms can sometimes cause power surges harmful to electronic equipment. For your own safety, take caution when using electric appliances during storms.

Cleaning your telephone

- Your VSP861 deskset has a durable plastic casing that should retain its luster for many years. Clean it only with a soft cloth slightly dampened with water or a mild soap.
- Do not use excess water or cleaning solvents of any kind.

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Remember that electrical appliances can cause serious injury if used when you are wet or standing in water. If the VSP861 deskset should fall into water, DO NOT RETRIEVE IT UNTIL YOU UNPLUG THE POWER CORD AND NETWORK CABLE FROM THE WALL, then pull the unit out by the unplugged cords.

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Appendix B: GPL License Information

Portions of the software associated with this product are open source, and fall within the scope of the GNU General Public License (GPL). Accordingly, those portions of code are available to the public, consistent with the requirements of the GPL, in either source code format or object code format, depending upon the nature of the code at issue. If you would like to exercise your right to receive the available code, please send a written request for the available code, along with a cashier's check, payable to VTech Communications, Inc., in the amount of \$15.00 (U.S.\$) to:

VTech Communications, Inc., 9590 SW Gemini Drive, Suite 120 Beaverton OR 97008

ATTN: Information Technology Group-VSP861 GPL code request

If your request does not fully comply with the foregoing requirements, VTech reserves the right to reject your request. Further, by requesting and receiving the available code, you release VTech, its affiliates, and its and their officers, directors, employees, and representatives ("VTech Parties") from any liability or responsibility relating to such code, and you acknowledge that the VTech Parties make no representations with respect to the origin, accuracy, usability, or usefulness of such code, and the VTech Parties have no responsibility to you whatsoever concerning the code, including without limitation any responsibility to provide explanation, support, upgrade, or any communication whatsoever. Your review or use of the available code is at your sole risk and responsibility.

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