



**VOIP-600 Series Phone**  
**Configuration and Operation Manual**

v1.0.1

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## Acronyms and Abbreviations

The following acronyms and abbreviations are commonly used throughout the document:

Acronyms	Definitions
ADA	Americans with Disabilities Act
AEC	Acoustic Echo Cancellation
AGC	Automatic Gain Control
ANSI	The American National Standards Institute a private non-profit organization that oversees the development of voluntary consensus standards for products, services, processes, systems, and personnel in the United States.
AUX	Auxiliary Input/Output. An Auxiliary Input accepts a contact closure from an external device, such as a Vehicle Detector, Door Switch, Scream Alert™, and card swipe. An auxiliary output produces a contact closure to an external device, such as a strobe light and motorized garage gate.
BABT	British Approvals Board of Telecommunications
CNG/VAD	Comfort Noise Generator/Voice Activity Detector. It is used to reduce the transmission rate during inactive speech periods while maintaining an acceptable level of output quality.
CSA	Canadian Standards Association
CE	The CE marking certifies that a product has met EU consumer safety, health or environmental requirements.
DHCP	Dynamic Host Configuration Protocol — protocol for assigning dynamic IP addresses to devices on a network.
DNS	Domain Name Server
DTMF	Dual Tone Multi Frequency signaling is used for telecommunication signaling over telephone lines.
FCC	Federal Communications Commission
FTP	File Transfer Protocol
GUI	Graphical User Interface
G.711	G.711 is codec also known as Pulse Code Modulation (PCM). It is the ITU-T international standard for encoding telephone audio on a 64 kbps channel.
G.723	G.723 is an ITU-T standard speech codec.
G.729	G.729 is an audio data compression algorithm. It is the ITU-T international standard for encoding telephone audio on 8 kbps channel.
IVR	Interactive Voice Response
IP-PBX	It is an IP based switch for call handling through public and private exchanges.
IE	Internet Explorer
IETF	The Internet Engineering Task Force (IETF) develops and promotes Internet standards.
PCBA	Printed Circuit Board Assembly
PoE	Power over Ethernet, IEEE 802.3af standard.

Acronyms	Definitions
QoS	Quality of Service is of particular concern for the continuous transmission of high-bandwidth video and multimedia information.
SIP	Session Initiation Protocol is a signaling protocol, widely used for setting up and tearing down multimedia communication sessions over network.
TIA	Telecommunications Industry Association
UL	Underwriters Laboratories
VoIP	Voice over Internet Protocol

## 1. Introduction

### 1.1. Overview of the Manual

This manual provides detailed instructions for the configuration and operation of VOIP-600 Series Phones. It is recommended to read this instructional manual completely before performing any configuration.

### 1.2. Intended Audience

This manual is targeted towards systems administrators, or any person who would configure and maintain VOIP-600 Series Phones. Fundamental knowledge in computer networking and Voice over Internet Protocol (VoIP) technologies is recommended for understanding this manual.

### 1.3. Objective

This manual provides a detailed examination of the features included in VOIP-600 Series Phones. It guides an administrator through the configuration and optimization of phone features. While configuration of the VOIP-600 Series Phones is covered in detail, configuration of other peripheral VoIP network elements is beyond the scope of this document.

### 1.4. Typographic Conventions

The following guidelines are used as typographic conventions in this user manual:

Item	Convention	Sample
Acronyms	All uppercase	SIP
Chapter titles	Title caps	See Chapter 3 Getting Started
Command-line commands and options (switches)	All lowercase, bold	<b>ifconfig</b> command <b>/a</b> option
Device names	All uppercase	VOIP-600
Directories	All lowercase	/flash
Error message names	Initial caps	Update failed
File names	Title caps (internal caps in short file names are acceptable for readability)	MainLogFile.txt BackupLogFile.txt
Menu names	Bold; title caps	<b>Insert</b> menu
Programs and applications	Usually title caps	<b>HyperTerminal</b>
Toolbar button names	Usually title caps (follow the interface); bold	<b>Apply</b> <b>Reset</b>
URLs	All lowercase; break long URLs before a forward slash, if necessary to break; do not hyphenate.	<a href="http://www.talkaphone.com/">http://www.talkaphone.com/</a>
User input	Usually title caps; bold	<b>Enter Password</b>

### 1.5. Related Documents

VOIP-600 Quick Installation Guide, v1.0

VOIP-600 Avaya IP Office Integration Guide, v1.0

VOIP-600 Cisco UCM Integration Guide, v1.0

VOIP-600 Alcatel-Lucent OmniPCX Integration Guide, v1.0

### 1.6. Technical Support / Help Desk

For technical assistance beyond the scope of this document, contact your distributor or Talkaphone Technical Support for further information.

Talk-A-Phone Co.  
7530 North Natchez Avenue  
Niles, Illinois 60714

Phone: 773.539.1100

Fax: 773.539.1241

E-mail: [support@talkaphone.com](mailto:support@talkaphone.com)

Web: [www.talkaphone.com](http://www.talkaphone.com)

## **2. Overview**

### **2.1. VOIP-600 Series Phone Overview**

The VOIP-600 series phones are outdoor-rated, ADA-compliant hands-free Voice over IP (VoIP) Emergency/Information Phones for use in locations such as: parking facilities, college campuses, medical centers, and industrial parks. VOIP-600 series phones are SIP<sup>1</sup> compliant.

The VOIP-600 series of phones are categorized into various device modes based on the number of buttons, keypad, handset and speaker/microphone present on the phone's faceplate.

These device modes are also given different marketing names. A model with "Keypad" is denoted with a letter "K" in its model number. A model with two buttons is denoted with a letter "D" in its model number. For example, VOIP-600 corresponds to "Single button Emergency VoIP phone". VOIP-600D is "Dual button Emergency VoIP phone". VOIP-600K is "Two-button Emergency VoIP Phone with Keypad".

Some of the functionality described below may not be available depending on the phone device mode. The rest of the document may use the term "phone" to refer to a "VOIP-600 series" phone.

NOTE: Not all features of the VOIP-600 series phones are available when used in conjunction with analog telephony adapter devices or with the Public Switched Telephone Network (PSTN).

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<sup>1</sup> RFC 3261 and RFC 2833 for DTMF delay



### 3. Getting Started

#### 3.1. Pre-requisites

Prior to configuring a VOIP-600 Series Phone, ensure the unit is powered on and connected to the network.

The VOIP-600 can be configured from a computer with either a TCP/IP network connection or a DB-9 Serial port. The VOIP-600 supports straight-through serial connections for basic programming. For access to the full configurable feature set, a modern web browser will be required.

The VOIP-600 currently supports access from:

- Internet Explorer 8 or greater
- Firefox 3.5 or greater

#### 3.2. Configuration Using the Web GUI

1. Ensure both the VOIP-600 and your PC are connected to the Local Area Network. A direct connection to the VOIP-600 will require the use of a crossover network cable.

VOIP-600 Series Phones are pre-configured with the following default settings:

**IP Address:** 192.168.1.10  
**Username:** admin  
**Password:** admin@123

2. Configure the IP address of your PC to be on the same subnet as the VOIP-600. For example, 192.168.1.3
3. Open a supported web browser and direct it to the IP address of the VOIP-600. For example, enter the following URL: <http://192.168.1.10>.

The browser prompts for authentication:

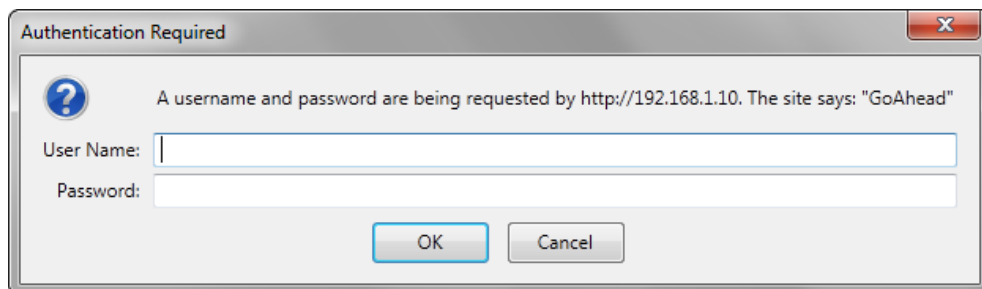


Figure 1 Authentication Required

4. Enter the default Username and Password. After authentication is successful, you are redirected to the Home page.
5. Further configuration of VOIP-600 settings are examined in Section 4: Using the Web GUI.

### 3.3. Configuration Using the Serial Console

Basic settings can be configured on VOIP-600 Series Phones through a serial console connection. Knowledge of Linux shell commands is recommended for serial console configuration.

Most commonly, the serial console connection can be used to set the IP address, as described below.

1. Connect a straight-through serial cable (DB9 Male to DB9 Female) from the serial port on the phone to an available COM port on the PC, noting which port was utilized.

2. Open your preferred serial console application (e.g., HyperTerminal or TeraTerm) on the PC and specify the following settings:

<b>Communications Port:</b>	<Determined in Step 1>
<b>Baud rate (bits per second):</b>	115200
<b>Data bits:</b>	8
<b>Parity:</b>	None
<b>Stop bits:</b>	1
<b>Flow control:</b>	None

3. Once connected, you will be presented with the following prompt:

```
root:/>
```

4. The /flash directory contains persistent files. It is highly recommended to switch to the /flash directory by entering the following command, and pressing **Enter**.

Note the space between 'cd' and '/flash':

```
root:/> cd /flash
```

5. Create a file where configuration parameters can be entered. For example, we create the config.txt file by entering the following command, and pressing **Enter**.

Note the space between 'vi' and 'config.txt':

```
root:flash/> vi config.txt
```

6. While in the vi application, press **i** to force vi into "Insert Mode."

7. The body of the config.txt file can now be created.

To set a new static IP address enter the following data:

```
NetworkMode = 3
```

```
IPAddress = <IP address, e.g. 192.168.1.10>
```

```
Netmask = <netmask, e.g. 255.255.255.0>
```

```
Gateway = <default gateway, e.g. 192.168.1.1>
```

8. Once the configuration options are entered, save the changes and exit vi by pressing **Escape**, then entering **:wq**

To exit without saving changes press **Escape**, then enter **:q!**

9. With the configuration file created, the final step is to apply these settings to the VOIP-600. Enter the following command, and press **Enter**.

```
root:flash/> configApp admin admin@123 config.txt
```

10. A reboot is required after serial console configuration. Enter the following command, and press **Enter**.

```
root:flash/> reboot
```

11. All configurable options are available in Appendix E: Sample Serial Configuration File.

## 4. Using the Web GUI

This section describes various configurable options available from the Web Interface of the phone. It is the primary interface for configuring and monitoring the phone. You must assign an IP address (Refer to 4.4.1 IP Settings) before configuring the features of the phone.

### 4.1. Web GUI Layout

This section describes the layout of the Web GUI.

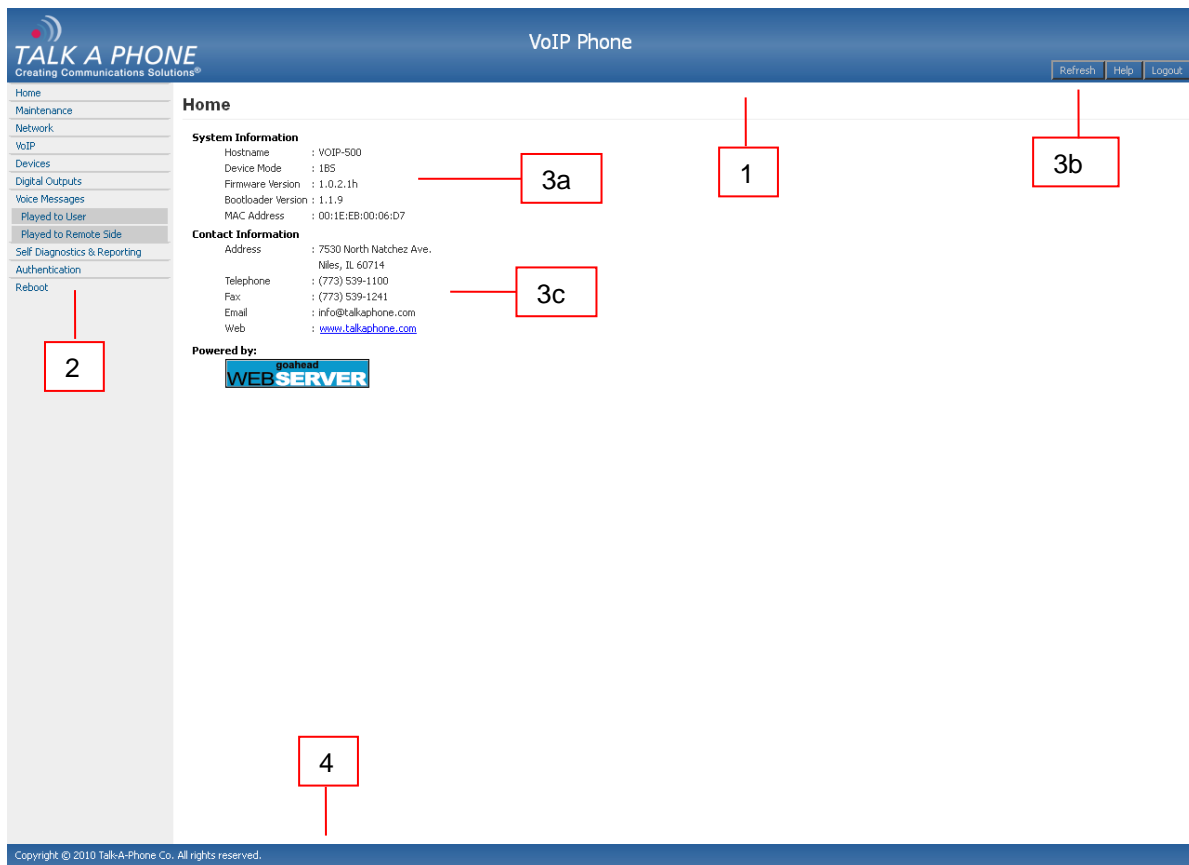



Figure 2 Using Web GUI

The Web GUI layout is divided into four sections:

1. **Header:** The Header section displays Talkaphone Logo and product title “VoIP Phone”. The right side of the header has the Refresh, Help, and Logout buttons. The Refresh button is used to reload the content section. The Help button provides help content from the Talkaphone website. The Logout button logs out the user. The Apply button will not be grayed out if the content pane (3) contains configurable parameters. The Apply button is used to save configuration changes to phone.
2. **Menu:** The menu page provides links to different menus and submenus available on the phone.
3. **Content:** The top part of the content section is used for Page Title (3a) and Status (3b) for any operation. Bottom part of content pane (3c) is used for displaying configurable fields when a menu or a submenu is selected. All the fields (text boxes/dropdown) display allowed field ranges.
4. **Footer:** The Footer section displays the Talkaphone copyright statement.

To update the phone configuration, the user must press the **Apply** button. After clicking **Apply** the status will be displayed in the “Status” section of the Content section (3b).

---

Note: When a configuration change is in the process of being applied, an  **Updating...** message appears.

An  **Updated.** message appears for the successful update.

An  **Update failed.** message appears if the input is incorrect. Incorrect inputs are marked with red.

---

**4.2. Web Application Menu**

This section provides a preview of the phone's Web Menu hierarchy/organization.

Home
Maintenance
Device State
Statistics
Logging
Date & Time
Ping & Traceroute
Firmware Upgrade
SMTP Server Configuration
Email Notification Profiles
Reset to Default
Network
IP Settings
SIP Settings
VoIP
Number Lists
Phone Settings
Audio Settings
Call Parameters
Paging Settings
Devices
Buttons
Auxiliary Inputs
LEDs
Auxiliary Outputs
Digital Outputs
Scripts
Events
Voice Messages
Played to User
Played to Remote Side
Self Diagnostics & Reporting
Authentication
Reboot

Figure 3 Application Menu

### 4.3. Home

The Home page displays phone information and contact information for Talkaphone.

#### Login → Home

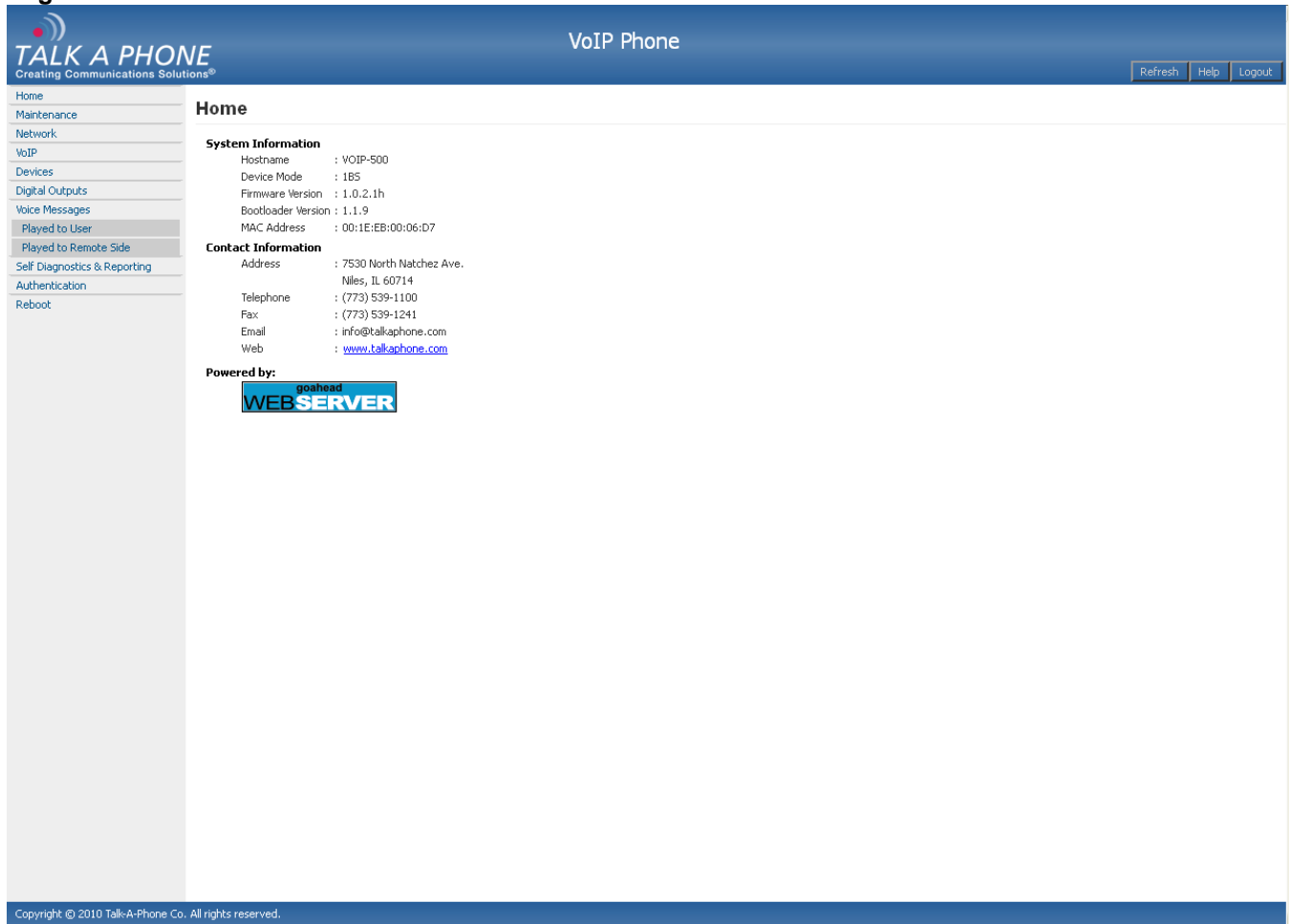


Figure 4 Home

Field Name	Description
Device Mode	This indicates the type of device. For example, 2BHSK indicates a device having two buttons, a handset, a keypad, and a speaker. Refer to Appendix D: VOIP-600 Device Modes for details on other device modes.
Firmware Version	Indicates the version of the phone's firmware
Bootloader version	Indicates the version of the bootloader that is used during startup (booting) of the phone. The bootloader version is independent of the firmware version.
MAC Address	Indicates the number assigned as the phone's unique MAC address.

#### 4.4. Network

##### 4.4.1. IP Settings

The IP Settings page allows you to configure the IP settings and the hostname of the phone.

The screenshot shows the 'IP Settings' page. At the top, there is a breadcrumb trail: 'Login → Network → IP Settings'. To the right of the breadcrumb are four buttons: 'Apply', 'Refresh', 'Help', and 'Logout'. The main heading is 'IP Settings'. Below it, the section 'Configure network connection :' has two radio buttons: 'DHCP - Automatic Configuration' and 'Static IP - Manual Configuration'. A red bracket labeled '1' groups these two options. Below this is the section 'Specify network details for "Static IP - Manual Configuration" :'. It contains four input fields: 'IP Address' (192.168.1.10), 'Subnet Mask' (255.255.255.0), 'Default Gateway' (192.168.1.1), and 'DNS Server'. A red bracket labeled '2' groups the IP Address and Subnet Mask fields. A red bracket labeled '3' groups the Default Gateway and DNS Server fields. Below these is the section 'Enter hostname :' with a 'Hostname' input field containing 'VOIP-500'. A red bracket labeled '4' groups this field. At the top right of the main content area, a red box labeled '5' points to the 'Apply' button.

Figure 5 IP Settings

To set IP settings:

1. Select **DHCP – Automatic Configuration** OR **Static IP - Manual Configuration**.
2. If **DHCP** is selected for configuration of the IP address, then the VOIP-600 should be rebooted after clicking 'Apply'.
3. Enter network details **IP Address, Subnet Mask, Default Gateway** of your network for **Static IP - Manual Configuration**.
4. Enter **DNS Server** to resolve domain names.
5. Enter **Hostname**. Use unique identifier for each device on the network.
6. Click **Apply** to save settings.



4.4.2. SIP Settings

This page is used to configure the phone number, SIP registrars and outbound proxy details.

The screenshot shows the 'SIP Settings' configuration page. At the top right, there are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. The page is titled 'SIP Settings' and contains several sections:

- Assign a phone number :** A text input field for 'Phone Number' containing '500'. A red box labeled '1' points to this field.
- Specify FQDN/IP Address :** Three text input fields for 'FQDN/IP Address' containing '192.168.40.227', 'voip500.local', and 'voip500.local'. A red box labeled '2' points to these fields.
- Enable/disable SIP registration :** A checkbox labeled 'Register' which is unchecked. A red box labeled '3' points to this checkbox.
- Specify SIP registrar :** A group of fields including 'Username' (500), 'Password' (masked with dots), 'IP Address' (192.168.40.88), 'Port' (5060), and 'Re-registration Time' (3600). A red box labeled '4' points to this entire section.
- Specify outbound proxy :** A group of fields including 'Username' (500), 'Password' (masked with dots), 'IP Address' (192.168.40.88), and 'Port' (5060). A red box labeled '5' points to this entire section.
- Registration status :** A radio button labeled 'Unregistered' which is selected.

A red box labeled '6' points to the 'Apply' button at the top right.

Figure 6 SIP Settings

To configure SIP settings:

1. Enter the **Phone Number** (number used to call from other SIP endpoints).
2. Enter the **Fully Qualified Domain Name** or the **IP Address** of the SIP Gateway/Proxy. Secondary and Tertiary FQDN or IP Addresses can be assigned for failover. There is a 30 character limit for this field. If Secondary and Tertiary Registrars are not used, then voip500.local is the default placeholder.
3. Check or Uncheck **Register** to enable or disable the SIP registration.
4. Enter **SIP registrar** details: (Secondary and Tertiary details can be assigned for failover)
  - **Username:** This is an optional field however it is required when Authentication is enabled on the SIP registrar. Valid entries include alphanumeric and special characters (\*, ', '-; No spaces allowed)
  - **Password:** This is an optional field however it is required when Authentication is enabled on the SIP registrar. Valid entries include alphanumeric and special characters (\*, ', '-; No spaces allowed)
  - **IP Address and Port:** Configure the IP Address and Port of the SIP registrar on which you want to register the phone. Default port is 5060. Valid port range is 1024-49151.
  - **Re-registration Time:** Configure the registration refresh time. On expiration of this time interval, the phone automatically sends a new register request to the registrar. Default re-registration time is 3600 seconds. Valid range is 10-14400 seconds.

5. Enter **Outbound Proxy** details: If a separate outbound proxy is not being used please enter the SIP registrar information again. (Secondary and Tertiary details can be assigned for failover)
  - **Username:** This is an optional field however it is required when Authentication is enabled on the SIP registrar. Valid entries include alphanumeric and special characters (\*, ', '-; No spaces allowed)
  - **Password:** This is an optional field however it is required when Authentication is enabled on the SIP registrar. Valid entries include alphanumeric and special characters (\*, ', '-; No spaces allowed)
  - **IP Address and Port:** Configure the IP Address and Port of the SIP registrar on which you want to register the phone. Default port is 5060. Valid port range is 1024-49151.
  - **Re-registration Time:** Configure the registration refresh time. On expiration of this time interval, the phone automatically sends a new register request to the registrar. Default re-registration time is 3600 seconds. Valid range is 10-14400 seconds.
  
6. Click **Apply**. The following type of registration status appears based on your selection:
  - Registering as <Phone Number> @ <Domain Name> - Registration is in progress with SIP registrar.
  - Registered as <Phone Number> @ <Domain Name> - Registration is completed and phone is able to receive incoming calls on the registered phone number.
  - Unregistered as <Phone Number> @ <Domain Name> - Phone is not registered for incoming calls. Outbound calls can still be established via outbound proxy.

## 4.5. VoIP

### 4.5.1. Number Lists

A phone can have seven lists of phone numbers. Each list can contain up to six numbers for round robin dialing. If the first number dialed is busy or does not answer after a specified number of rings, the phone dials the next number in the phone number list. The phone continues trying the numbers in the list in order until it reaches the last number programmed in the list. If the last phone number is also busy or does not answer after the specified number of rings, the phone should go back to dialing the first number, then the second number, etc. until the call is completed, or until the time limit defined by Call Conversation Timer expires.

The numbers from these lists are dialed on call initiation when the associated buttons are pressed or when the associated Auxiliary Inputs are activated (Refer to 4.6.5 Auxiliary Inputs).

Figure 7 Number Lists

To set numbers in the number lists:

1. Use the drop-down menu to select the number list to be configured (List1 is the default list assigned to Button 1.)
2. Provide up to six numbers. Numbers can be in the format of plain numbers/aliases or as number@ip address. Valid range is up to 30 characters.
3. Click **Apply** to save settings.

4.5.2. Phone Settings

The Phone Settings page allows you to configure the VoIP protocol for communication, auto answer mode of operation, prioritization between phone and paging operation, and caller ID settings.

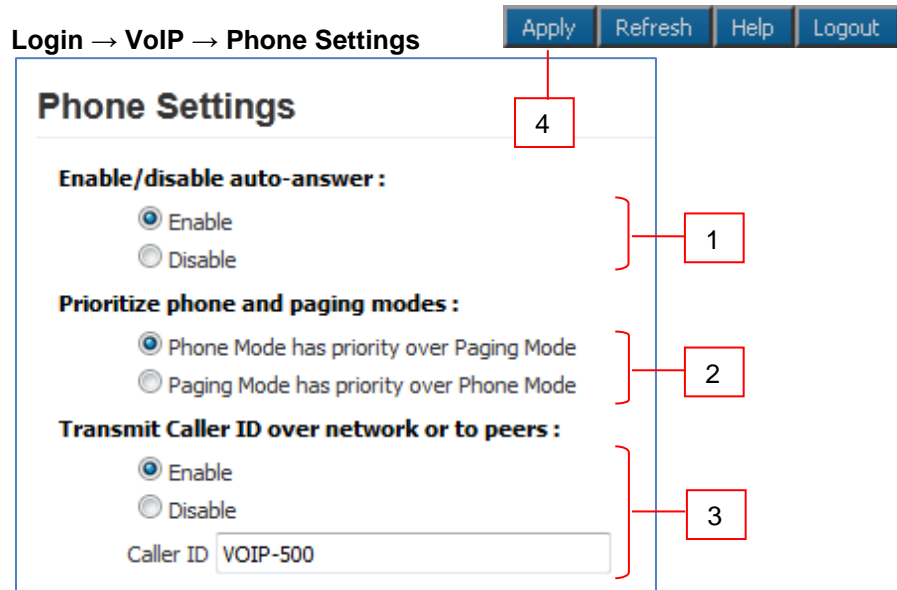


Figure 8 Phone Settings

To configure the phone settings:

1. Select **Enable** if you want the VOIP-600 to answer any incoming calls automatically (only applicable to VOIP-600 phones with a built-in hands-free speaker).

When this option is set to **Disable**, incoming calls are answered by either picking up the handset or by pressing any button configured as a hook switch. If the VOIP-600 does not have a handset option and cannot be configured with a hook switch button, a manual answer method will not be available.

2. Select and assign the priority between **Phone Mode** and **Paging Mode**. When the “phone mode” has priority over the “paging mode,” the incoming paging request is processed when the phone is in idle state. When the phone mode has a higher priority, all paging operations stop and phone operations are processed. When the paging mode has a higher priority, all phone operations stop when the paging mode has a higher priority and the paging request is processed.
3. Select **Enable** or **Disable** to transmit a **Caller ID** over the network or to peers. This field allows alphanumeric characters only. Some phone systems require Caller ID assignment from within the phone system management.
4. Click **Apply** to save settings.

4.5.3. Audio Settings

The Audio Settings page allows you to configure settings for preferred VoIP codec, enable/disable audio processing modules, outbound DTMF duration, line level output gain, and the microphone and speaker parameters.

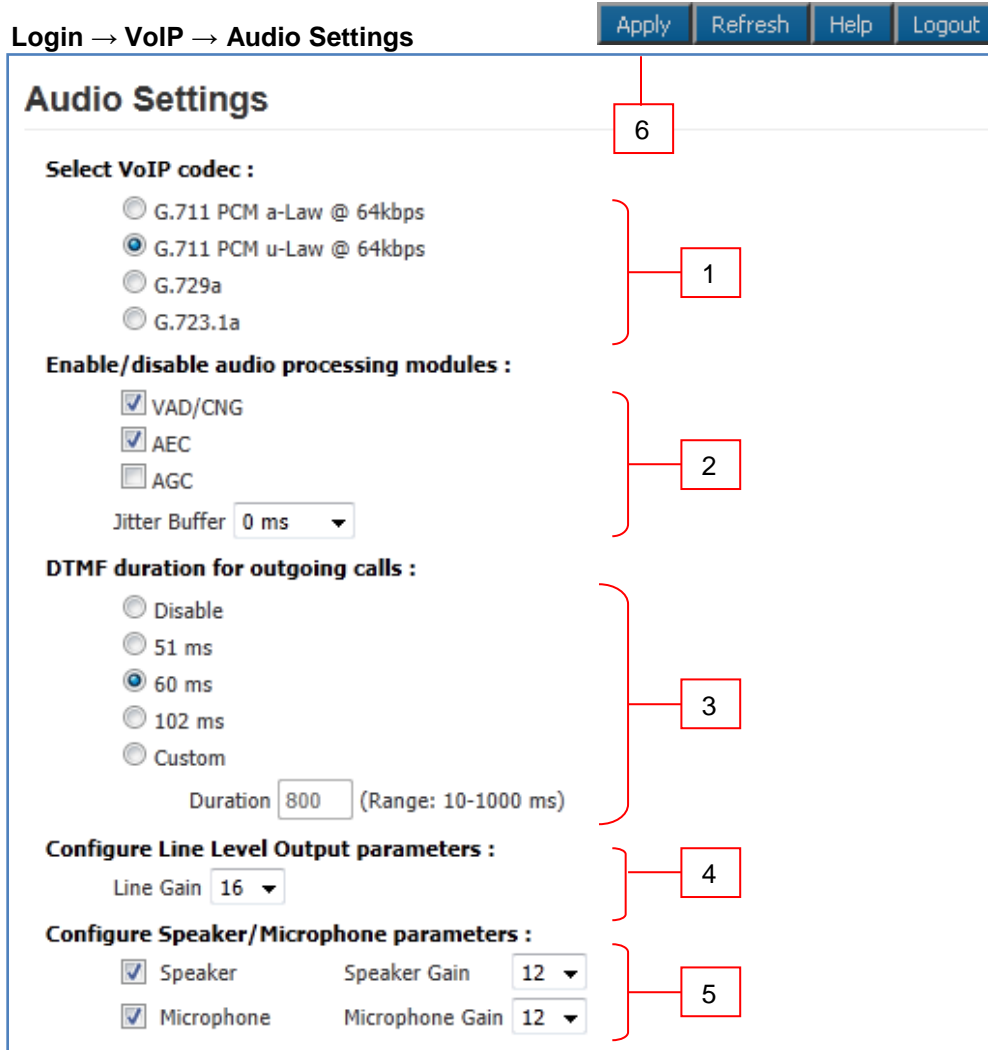


Figure 9 Audio Settings

To configure audio settings:

1. Select **G.711 PCM a-Law @ 64kbps**, **G.711 PCM u-Law @ 64kbps**, **G.729A** or **G.723.1a** for the preferred VoIP codec. The selected codec is used for outgoing call negotiation. (Default is **G.711 PCM u-Law @ 64kbps**.)

2. Check or uncheck **VAD/CNG**, **AEC**, or **AGC** to enable/disable the audio processing modules.
  - **VAD/CNG** (Voice Activity Detection / Comfort Noise Generation) is enabled by default. VAD/CNG is used to reduce background noise and also produces as low but audible comfort noise.
  - **AEC** (Acoustic Echo Cancellation) is enabled by default. AEC is used to remove the echo from voice communications in a speakerphone environment.
  - **AGC** (Automatic Gain Control) is disabled by default. AGC can be used where audio input levels vary widely in an effort to enhance/reduce the audio input to the phone.
3. Select the Outbound DTMF<sup>2</sup> duration of **51 msec**, **60 msec**, **102 msec**, or **Custom** to be used for sending DTMF packets during a call. Select **Disable** if Outbound DTMF is not required. This configuration option is only available for a device mode with a “K” designator (e.g., 1BKH, 4BHSK).
4. Click the drop-down menu to select **Line Gain** for the Line Level Output parameter. Each level increases/decreases the line gain in steps of approximately 3dB.
5. Enable or disable the **Speaker** and/or **Microphone**. Configuration changes made to the Speaker parameters will affect the built-in hands-free speaker and the optional handset speaker. Configuration changes made to the Microphone parameters will affect the built-in hands-free microphone and the optional handset microphone.
6. The **Speaker Gain** and the **Microphone Gain** can be individually adjusted. Each level increases/decreases gain in steps of approximately 3dB.
7. During an active call, the Speaker/Microphone gains can be adjusted during an active call using DTMF Operation codes (Refer to Appendix A: Remote Operation Key Codes – Quick Reference: Speaker Enable/Disable and Microphone Enable/Disable). Gain adjustments made via DTMF only takes effect for the instance of the phone call where the DTMF Operation Code was applied.
8. Click **Apply** to save settings.

---

<sup>2</sup> Out-of-band RTP

4.5.4. Call Parameters

The Call Parameters page is used to configure the mandatory and the optional call related parameters such as call progress tones, answer/disconnect key, welcome tone, and timers.

The screenshot shows the 'Call Parameters' configuration page. At the top right, there are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. A red box labeled '6' points to the 'Apply' button. The main content area is titled 'Call Parameters' and contains several sections:

- Enable/disable call progress tones :** Includes radio buttons for 'Enable' (selected) and 'Disable'. A red box labeled '1' points to this section.
- Specify key to answer and/or disconnect a call from the Remote Side :** Includes two dropdown menus: 'To disconnect a call,press # key' and 'To answer a call,press Disable'. A red box labeled '2' points to these dropdowns.
- Enable/disable "Welcome Tone" :** Includes radio buttons for 'Enable' (selected) and 'Disable'. A red box labeled '3' points to this section.
- Configure required timers :** Includes five input fields with ranges: Provisional Timer (5, Range: 5-20 sec.), Ringer Timer (5, Range: 1-12 rings), Hang-up Timer (0.5, Range: 0.5-3.0 sec.), Local Interdigit Timer (5, Range: 5-20 sec.), and Remote Interdigit timer (5, Range: 5-20 sec.). A red box labeled '4' points to this entire section.
- Configure optional timers :** Includes three checked checkboxes with input fields and ranges: Call conversation Timer (12, Range: 1-360 min.), Ringback or Busy Timer (15, Range: 1-60 sec.), and Hang-up On Silence Timer (30, Range: 10-360 secs). A red box labeled '5' points to this section.

Figure 10 Call Parameters

To set the call parameters:

1. Select **Enable/Disable** to allow/mute call progress tones to be heard by the user until the call is answered by the remote side.
2. Click the drop-down menu to set a key to answer a call and/or disconnect a call from the remote side. You can assign a key within the range of 0-9, \*, #, or Disable.
3. Select **Enable** or **Disable** to enable or disable the "Welcome Tone". When enabled, a tone is played to the remote side when a call is placed to the VOIP-600 phone that auto-answers the call. The "Welcome Tone" assists with distinguishing ETP-400 Series Phones from the VOIP-600 Series Phones.

4. Configure required timers:
  - **Provisional Timer** is the time interval for which the phone waits for the call initiation acknowledgement from the remote side. This timer is started when the call is initiated. This timer is useful in situations where the remote phone is unreachable for unknown reasons. The default value is set to five seconds. This timer is used only during round robin dialing.
  - **Ringer Timer** is the number of rings (one ring = six seconds) the phone waits before dialing the next number. The default value is five rings. This timer is used only during round robin dialing.
  - **Hang-up Timer** is the time interval for which the phone waits before initiating a new call after hanging up the current call. The default value is 0.5 seconds.
  - **Local Inter-digit Timer** is the time interval between two consecutive dialed digits. The inter-digit timer is started when the first digit is entered and is restarted after each new digit is entered. When the user dials a number, a call gets initiated after the expiry of this timer. This timer also applies during local key code translations for performing the operations. The default value is five seconds.
  - **Remote Inter-digit Timer** is the time interval for which the phone is expecting a digit to be received from remote side during key press to answer and for DTMF Operation Codes<sup>3</sup> that require remote key codes for activation/deactivation. The default value is five seconds.
5. Configure optional timers:
  - **Call Conversation Timer** is the time interval after which the call is terminated automatically. The default value is 12 minutes. If the call conversation time is disabled the conversation lasts for an indefinite duration.
  - **Ringback or Busy Timer** is the time interval for which the phone waits on the call queue before dialing the next number from the number list. The default value is 15 seconds. The phone waits for indefinite time if this timer is disabled. This timer is used only during round robin dialing.
  - **Hang-up on Silence Timer** is the time interval after which a call is terminated automatically if the voice or audio channel is idle. The default value is 30 seconds.
6. Click **Apply** to save settings.

---

<sup>3</sup> For example, the remote key code sequence 123 is used for activating Aux output #1. During a call, the remote side presses 12, now the phone is expecting key 3 to be received before the expiry of this timer, if key 3 is pressed before expiry of this timer, a single beep is heard at the remote side followed by Aux Output #1 Activation at the local end. If not, the phone considers this as an unsuccessful operation and will play two beeps to the remote side.



4.5.5. Paging Settings

The VOIP-600 Series Phone can be used as a paging device by a WEBS Contact® Server. The page can be output through either the built-in hands-free speaker or the line level output.

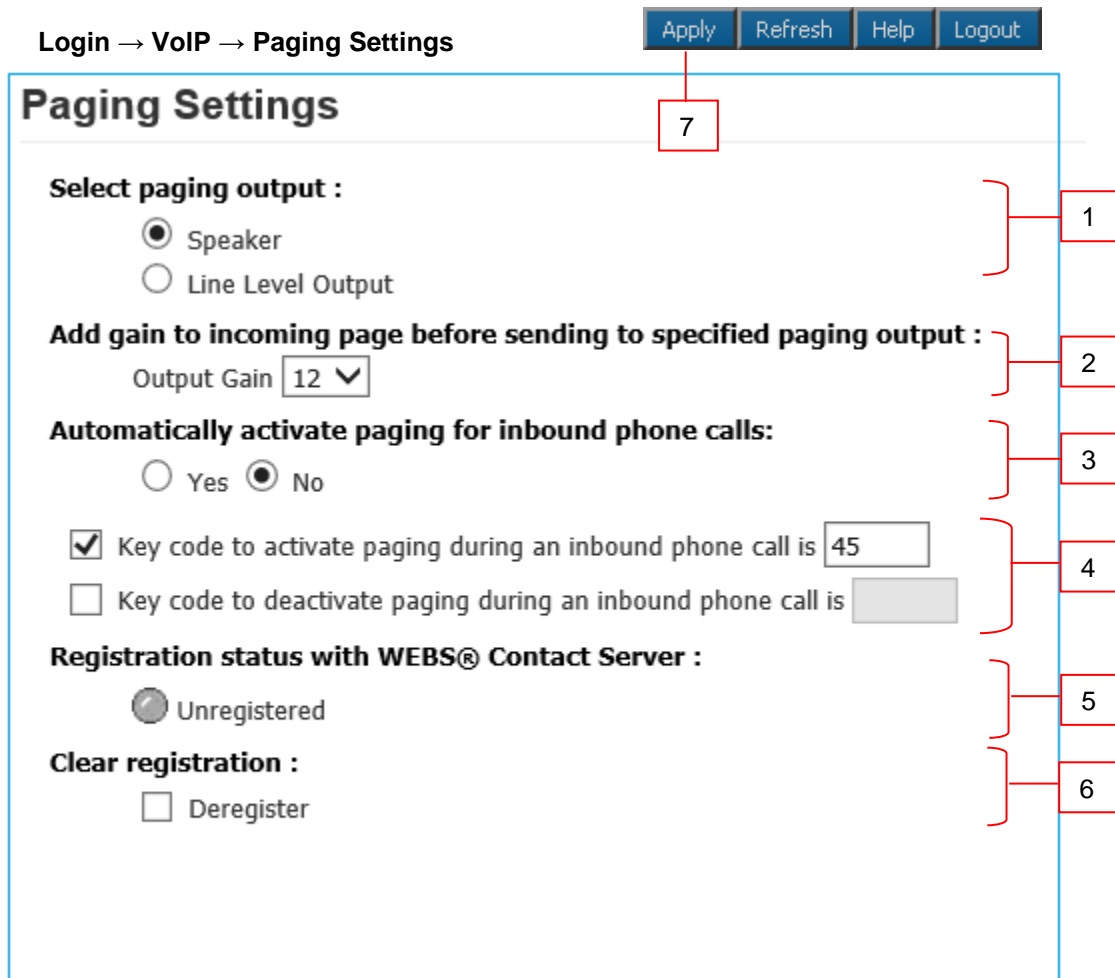


Figure 11 Paging Settings

To configure paging settings:

1. Select **Speaker** or **Line Level Output** to send the incoming audio page to the built-in hands-free speaker or the line level output. The speaker option is only available for device modes with an “S” designator (e.g., 1BS, 4BHSK).
2. Click the drop-down menu to adjust the **Output Gain** for all incoming pages before sending to the selected audio paging output. Each level adjustment corresponds to approximately 3dB.
3. Click ‘Yes’ to **Automatically activate paging** for inbound phone calls to the VOIP-600.
4. Enter the Keycodes **to activate and deactivate paging** during an inbound phone call to the VOIP-600.
5. The phone displays Unregistered or Registered @ <WEBS\_Contact\_Server\_IP\_Address> for Registration Status on the WEBS Contact® Server.

6. If it deregistration is required from the WEBS Contact® Server in error situations where the WEBS Contact® Server is unable to deregister the device, select **Deregister** check box to clear the registration status.
7. Click **Apply** to save settings.

**4.6. Devices**

**4.6.1. Buttons**

The Buttons configuration page allows for programming of the buttons depending on the device mode.

The screenshot shows the 'Buttons' configuration page with the following settings:

- Buttons** (Section Header)
- Configure Button #1 :**
  - Button #1 Mode: Always Autodial
  - Call from Number List: List 1 (Callout 2(i))
  - Call Priority: 1
  - Network Priority: 46 (Range: 0-63) (Callout 2(iii))
- Configure Button #2 :**
  - Button #2 Mode: Autodial
  - Call from Number List: List 1
  - Call Priority: 2 (Callout 2(ii))
  - Network Priority: 46 (Range: 0-63)
- Configure Button #3 :**
  - Button #3 Mode: Volume Down (Callout 1)
  - Call from Number List: List 1
  - Call Priority: 2
  - Network Priority: 46 (Range: 0-63)
- Configure Button #4 :**
  - Button #4 Mode: Volume Up
  - Call from Number List: List 1
  - Call Priority: 2
  - Network Priority: 46 (Range: 0-63)

Navigation buttons: Apply, Refresh, Help, Logout

Figure 12 Buttons

To configure buttons:

1. Select the button mode for each button you want to configure. The mode for Button #1 is not configurable and is always set to **Autodial**.

Buttons #2, #3, and #4 can be configured to **Autodial** or **Hook-switch**. Optionally, Buttons #3 and #4 can be configured as **Volume Up/Volume Down/Play Voice Message**. The number of buttons available for configuration will be based on the Device Mode (e.g., Device Mode 2BS will have Button #1 and #2 available for configuration).

2. When a Button is configured for **Autodial** mode, configure the following:
  - i. Use the drop-down menu to assign the Number List. Phone Numbers from the Number List are dialed in round robin fashion when the corresponding button is pressed.
  - ii. Select the **Call Priority**. This parameter assigns a priority to the call that is generated when the corresponding button is pressed. A call with a “Call Priority” of 1 terminates all the existing calls having the call priority in the range of 2–10.
  - iii. Select the **Network Priority**<sup>4</sup> to assign a network priority to the placed call. The value is utilized by the network to prioritize voice traffic. The default value is 46.
3. Click **Apply** to save settings.

---

**Note** Button #1 has the highest call priority and this cannot be changed. Pressing Button #1 cancels all operations and places a call.

---

---

<sup>4</sup> Network priority is the same as Differentiated Services Code Point (DSCP). Do not change this unless absolutely required as it might degrade the voice quality of the phone.

4.6.2. Hook Switch

The Hook switch configuration page configures the numbers to be dialed when the handset goes off-hook. This configuration page is only available for the 0BH device mode.

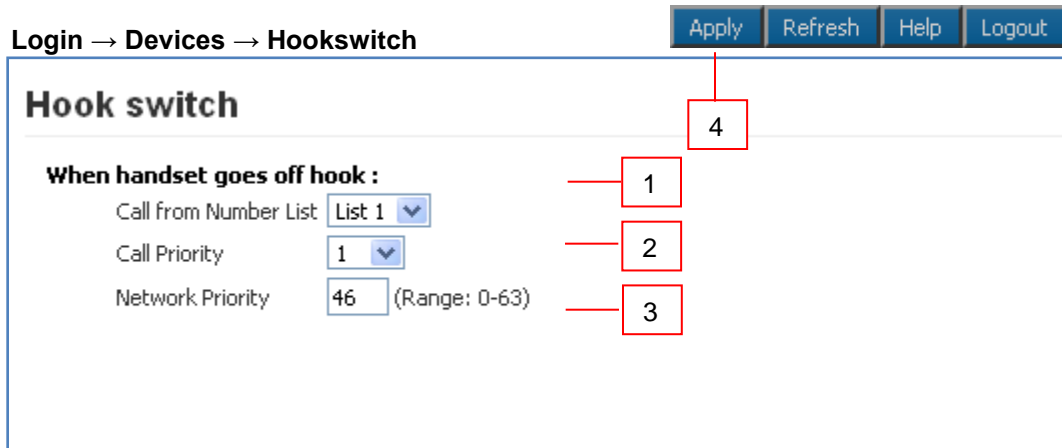


Figure 13 Hookswitch

To configure the Hook switch:

1. Click the drop-down menu to assign a **Number List**. Phone Numbers from the Number List are dialed in round robin fashion when the handset goes off-hook.
2. Select the **Call Priority**. This parameter assigns a priority to the call that is generated when the handset goes off-hook. A call with a “Call Priority” of 1 terminates all the existing calls having the call priority in the range of 2–10.
3. Select the **Network Priority**<sup>5</sup> to assign a network priority to the placed call. The value is utilized by the network to prioritize the voice traffic. The default value is 46.
4. Click **Apply** to save settings.

<sup>5</sup> Network priority is the same as Differentiated Services Code Point (DSCP). Do not change this unless absolutely required as it might degrade the voice quality of the phone.

4.6.3. Keypad

The Keypad configuration page allows administrator to configure the keypad of the phone for “normal” or “speed dial” mode. This configuration page is only available for phones with a keypad.

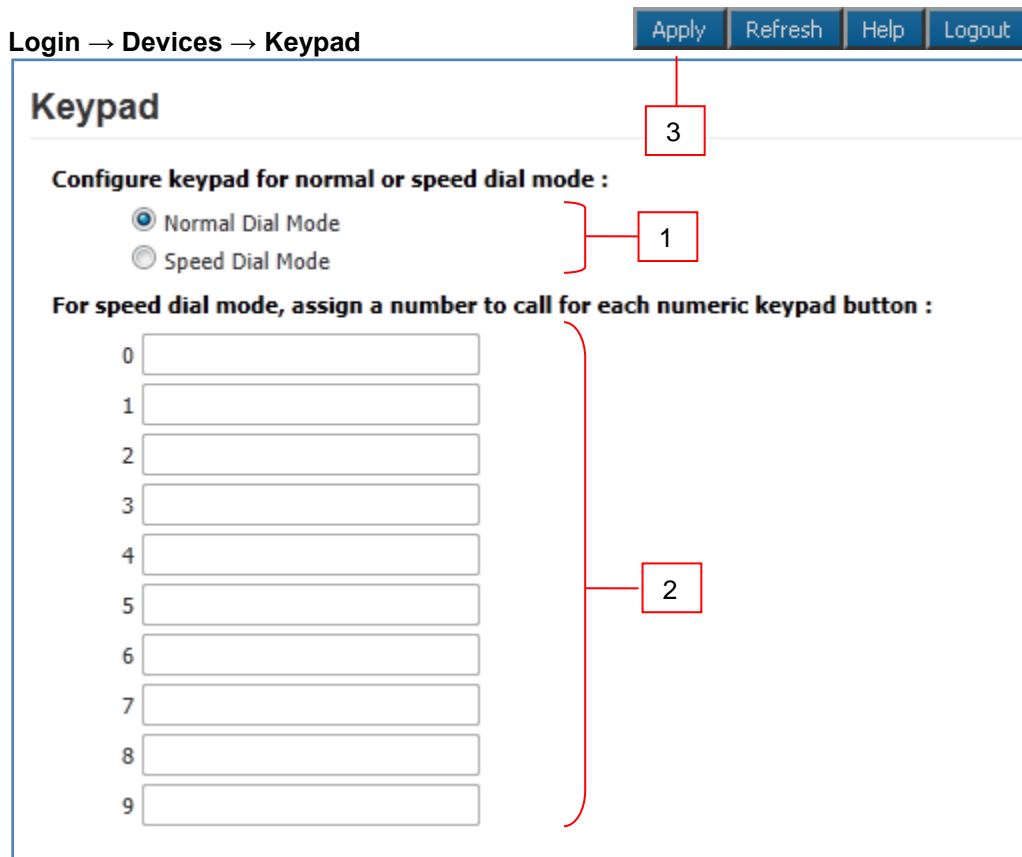


Figure 14 Keypad – Normal Dial Mode

To configure the Keypad:

1. Select the dial mode – **Normal Dial Mode** or **Speed Dial Mode**. Normal Dial Mode allows for manual dialing of an “analog style” phone number. If the phone is off-hook, dialing numbers on the keypad will place a call after the local inter-digit timer expires (Refer to 4.5.4 Call Parameters: Local Inter-digit Timer).
2. For Speed Dial Mode, enter the number for each keypad button (0–9). The number can be in the format of an analog phone number, an IP address, an alias, or a number@domain. When the phone is taken off-hook, a call can be placed directly to the number by pressing the corresponding keypad buttons.
3. Click **Apply** to save settings.

4.6.4. LEDs

The LEDs configuration page allows for the configuration of LED #3.

**Login → Devices → LEDs** Apply Refresh Help Logout

**LEDs**

**Activate LED #3 when :**

Keycode entered from remote side is  for  duration  seconds.

**Deactivate LED #3 when :**

Keycode entered from remote side is

**Note**  
Duration is in the range: 1-3600

Figure 15 LED Settings

To configure the LEDs:

1. Enter the **Key code sequence** and **duration** for LED #3 to be activated.
  - i. Key code sequence to be used by the remote side to activate LED #3<sup>6</sup>.
  - ii. Click the drop-down menu to select the duration for which LED #3 should remain active (i.e. ENTIRE CALL (deactivates after the termination of the call), CONTINUOUS (activated indefinitely until deactivated by deactivation event), or CUSTOM (activated for the specified duration in seconds).
2. Enter the Key code sequence to be used by the remote side to deactivate LED #3<sup>7</sup>.
3. Click **Apply** to save settings.

<sup>6</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

<sup>7</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

**4.6.5. Auxiliary Inputs**

The Auxiliary Inputs configuration page configures the numbers to be dialed on activation of Auxiliary Input devices.

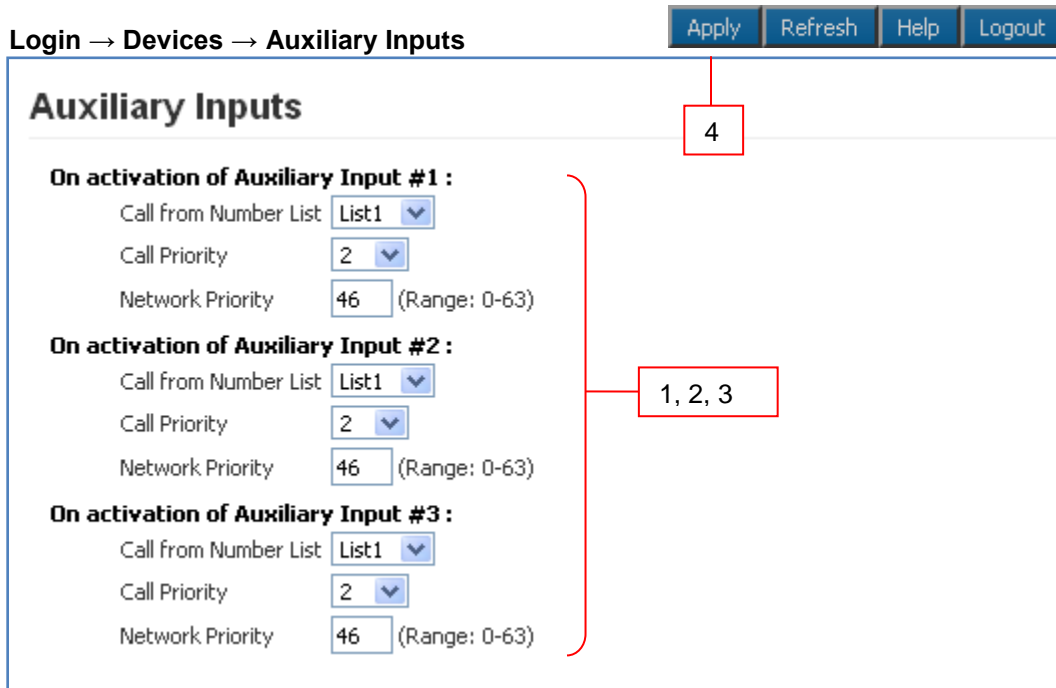


Figure 16 Auxiliary Inputs

To configure the auxiliary inputs:

- 1 Click the drop-down menu to assign the Number List. Numbers from the Number List are dialed in round robin fashion when the corresponding Auxiliary Input is activated. Select **None** if this feature is not desired upon activation of an Auxiliary Input.
- 2 Select the **Call Priority**. This parameter assigns a priority to the call that is generated upon activation of the corresponding Auxiliary Input. A call with a “Call Priority” of 1 terminates all the existing calls having the call priority in the range of 2–10.
- 3 Select the **Network Priority**<sup>8</sup> to assign a network priority to the placed call. The value is utilized by the network to prioritize the voice traffic. The default value is 46.
- 4 Click **Apply** to save settings.

<sup>8</sup> Network priority is the same as Differentiated Services Code Point (DSCP). Do not change this unless absolutely required as it might degrade the voice quality of the phone.



4.6.6. Auxiliary Outputs

The Auxiliary Output page allows events to activate/deactivate the auxiliary output device.

Login → Devices → Auxiliary Outputs

Apply Refresh Help Logout

### Auxiliary Outputs

Select Auxiliary Output **1** and assign a label of **Aux Label 1**

**Activate Auxiliary Output upon events :**

**Button Events**

Call with Button #1 is **INITIATED** for **ENTIRE CALL** with duration **10** sec.

**Auxiliary Input Events**

Auxiliary Input 1 is activated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

Auxiliary Input 1 is deactivated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

Auxiliary Input 2 is activated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

Auxiliary Input 2 is deactivated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

Auxiliary Input 3 is activated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

Auxiliary Input 3 is deactivated during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

**Remote Side Key Codes**

Key code **\*11\*** is entered by Remote Side for **ENTIRE CALL** with duration **10** sec.

**Local Key Codes**

Key code **[ ]** is entered by User during :  
Active call state for **ENTIRE CALL** with duration of **10** sec.  
Idle state for **CONTINUOUS** with duration of **10** sec.

**Deactivate Auxiliary Output upon events :**

**Button Events**

Call with Button #1 is INITIATED

**Auxiliary Input Events**

Auxiliary Input 1 is activated

Auxiliary Input 1 is deactivated

Auxiliary Input 2 is activated

Auxiliary Input 2 is deactivated

Auxiliary Input 3 is activated

Auxiliary Input 3 is deactivated

**Remote side Key Codes**

Key code \*21\* is entered by Remote Side

**Local Key Codes**

Key code [ ] is entered by User

**On Selected Events**

On completion of selected deactivation events

**On Call Transfer**

On Call Transfer

Figure 17 Auxiliary Outputs

To configure the auxiliary outputs:

1. Use the drop-down menu to select the Auxiliary Output. If desired, assign a label to the Auxiliary Output.
2. Configure various events that should activate an Auxiliary Output.

i. **Button Events**

The button events are available only when the Buttons are programmed for Autodial mode (Refer to 4.6.1 Buttons: for **Autodial**).

- Configure Auxiliary Output activation on Button #1 and/or Button #2 when a call is **INITIATED**, **ANSWERED**, or **TERMINATED**.
- Use the drop-down menu to select the duration for which the Auxiliary Output should remain active (i.e. **ENTIRE CALL** (deactivates after the termination of the call), **CONTINUOUS** (activated indefinitely until deactivated by a deactivation event), or **CUSTOM** (activated for the specified duration in seconds)).

ii. **Auxiliary Input Events**

- Configure the Auxiliary Output activation on activation/deactivation of Auxiliary Input #1, Auxiliary Input #2, and Auxiliary Input #3.
- Configure the duration separately for the idle state and the active call state. Use the drop-down menu to select the duration for which the Auxiliary Output should remain active (i.e. **ENTIRE CALL** (deactivates after the termination of the call), **CONTINUOUS** (activated indefinitely until deactivated by a deactivation event), or **CUSTOM** (activated for the specified duration in seconds)).

- iii. **Remote Side Key Codes**
    - Configure the Key Code sequence to be used by the remote side to activate the corresponding Auxiliary Output.
    - Use the drop-down menu to select the duration for which the Auxiliary Output should remain active (i.e. **ENTIRE CALL** (deactivates after the termination of the call), **CONTINUOUS** (activated indefinitely until deactivated by a deactivation event), or **CUSTOM** (activated for the specified duration in seconds)).
  - iv. **Local Key Codes**
    - Configure the Key Code sequence to be used by the Local User to activate the corresponding Aux Output device<sup>9</sup>.
    - Use the drop-down menu to select the duration for which the Auxiliary Output should remain active (i.e. **ENTIRE CALL** (deactivates after the termination of the call), **CONTINUOUS** (activated indefinitely until deactivated by a deactivation event), or **CUSTOM** (activated for the specified duration in seconds)). Configure the Key Code sequence to be used by the remote side to activate the corresponding Auxiliary Output.
3. Configure various events on which the Auxiliary Output(s) needs to be deactivated.
- i. **Button Events**

The button events are available only when the Buttons are programmed for Autodial mode (Refer to 4.6.1 Buttons: for **Autodial**).

    - Configure the Auxiliary Output deactivation on Button #1 and/or Button #2 when a call is **INITIATED, ANSWERED, or TERMINATED**.
  - ii. **Auxiliary Input Events**
    - Configure the Auxiliary Output deactivation on activation/deactivation of Auxiliary Input #1, Auxiliary Input #2, and Auxiliary Input #3.
  - iii. **Remote Side Key Codes**
    - Configure the Key Code sequence to be used by the remote side to for deactivation of the corresponding Auxiliary Output<sup>10</sup>.
  - iv. **Local Key Codes**
    - Configure the Key Code sequence to be used by the Local User for deactivation of the corresponding Auxiliary Output.
  - v. **On Selected Events**
    - Configure the Auxiliary Output deactivation on selected events as specified above.
  - vi. **On Call Transfer**
    - Configure the Auxiliary Output deactivation on call transfer.
4. Click **Apply** to save settings.

---

**Note** You cannot use the same event for activation and deactivation of the same Auxiliary Output.

---

<sup>9</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

<sup>10</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

## 4.7. Digital Outputs

### 4.7.1. Digital Output Scripts

The VOIP-600 supports ASCII digital outputs to third-party devices. The Digital Outputs – Scripts page allows for creation of scripts that will run when activated on the Digital Outputs – Events page.

Figure 18 Digital Outputs - Scripts

To create Digital Output Scripts:

1. Use the drop-down menu to select the **Digital Output Script slot**, and assign a **label**.
2. Enter a script to be run as follows:
  - i. The entire script cannot exceed 300 characters.
  - ii. The scripting engine supports the following commands:
    - i. **echo** This command is used to declare the message content
    - ii. **nc** This command is used to declare which server to communicate with
    - iii. **|** The pipe symbol is used to pass the output of one command to the input of another
    - iv. **+** The plus symbol is used to separate multiple command sets
  - iii. The **echo** command is used to declare the message content.  
 Proper usage is: `echo "message content"`  
 Example: `echo "activate cam_1"`
  - iv. The **nc** (netcat) command is used to declare which server to communicate with.  
 Proper usage is: `nc <IP_ADDRESS> <PORT>`  
 Example: `nc 192.168.1.100 23`
  - v. The **|** (pipe) symbol is used to pass the output of one command to the input of another. In this context, we utilized the pipe to pass the message content (echo) to the server connection (nc).  
 Example: `echo "activate cam_1" | nc 192.168.1.100 23`
  - vi. The **+** (plus) symbol is used to separate multiple command sets.

Example:                   echo "activate cam\_1" | nc 192.168.1.100 23 +  
                              echo "activate cam\_2" | nc 192.168.1.100 23

3. Click **Apply** to save settings.

Note:

The digital output scripts can be utilized to perform HTTP GET and POST requests in addition to basic ASCII output. As static message content is declared with the echo command, only HTTP Basic Authentication is supported.

HTTP requests require line breaks requiring the echo command to interpret backslash escapes. The `-e` options enables this functionality.

Also note that all HTTP headers will need to be manually specified.

The full HTTP specification is outside the scope of this document, however an example GET and POST script are shown below for illustration.

Example GET:

```
echo -e "GET /axis-cgi/com/ptz.cgi?gotoserverpresetname=POSITION1 HTTP/1.1\nHost: 192.168.1.100\nAuthorization: Basic YWRtaW46YWRtaW4=\n" | nc 192.168.1.100 80
```

Example POST:

```
echo -e "POST /api/static-device/alert/ HTTP/1.1\nHost: 192.168.1.100\nAuthorization: Basic YWRtaW46YWRtaW4=\nntype=VOIP-600&description=Activate_1\n" | nc 192.168.1.100 80
```

**4.7.2. Digital Output Events**

The Digital Output Events page allows events to activate the digital output of scripts.

[Login](#) → [Digital Outputs](#) → [Events](#)

[Apply](#) [Refresh](#) [Help](#) [Logout](#)

### Digital Outputs - Events

**Primary Digital Output events :**

**Button Events**

Call with Button #1 is INITIATED execute Script 1

**Auxiliary Input Events**

Auxiliary Input 1 is activated execute Script 1

Auxiliary Input 1 is deactivated execute Script 1

Auxiliary Input 2 is activated execute Script 1

Auxiliary Input 2 is deactivated execute Script 1

Auxiliary Input 3 is activated execute Script 1

Auxiliary Input 3 is deactivated execute Script 1

**Remote Side Key Codes**

Key code [ ] is entered by Remote Side. execute Script 1

**Local Key Codes**

Key code [ ] is entered by User execute Script 1

**Secondary Digital Output events :**

**Button Events**

Call with Button #1 is INITIATED after a duration of 10 seconds, execute Script 1

**Auxiliary Input Events**

Auxiliary Input 1 is activated during :

Active call state: after a duration of 10 seconds, execute Script 1

Idle state: after a duration of 10 seconds, execute Script 1

Auxiliary Input 1 is deactivated during :

Active call state: after a duration of 10 seconds, execute Script 1

Idle state: after a duration of 10 seconds, execute Script 1

Auxiliary Input 2 is activated during :

Active call state: after a duration of 10 seconds, execute Script 1

Idle state: after a duration of 10 seconds, execute Script 1

Auxiliary Input 2 is deactivated during :

Active call state: after a duration of 10 seconds, execute Script 1

Idle state: after a duration of 10 seconds, execute Script 1

Auxiliary Input 3 is activated during :

Active call state: after a duration of 10 seconds, execute Script 1

Idle state: after a duration of 10 seconds, execute Script 1

Callout 4 points to the top navigation bar (Login → Digital Outputs → Events).

Callout 1 points to the Primary Digital Output events section.

Callout 2 points to the Secondary Digital Output events section.

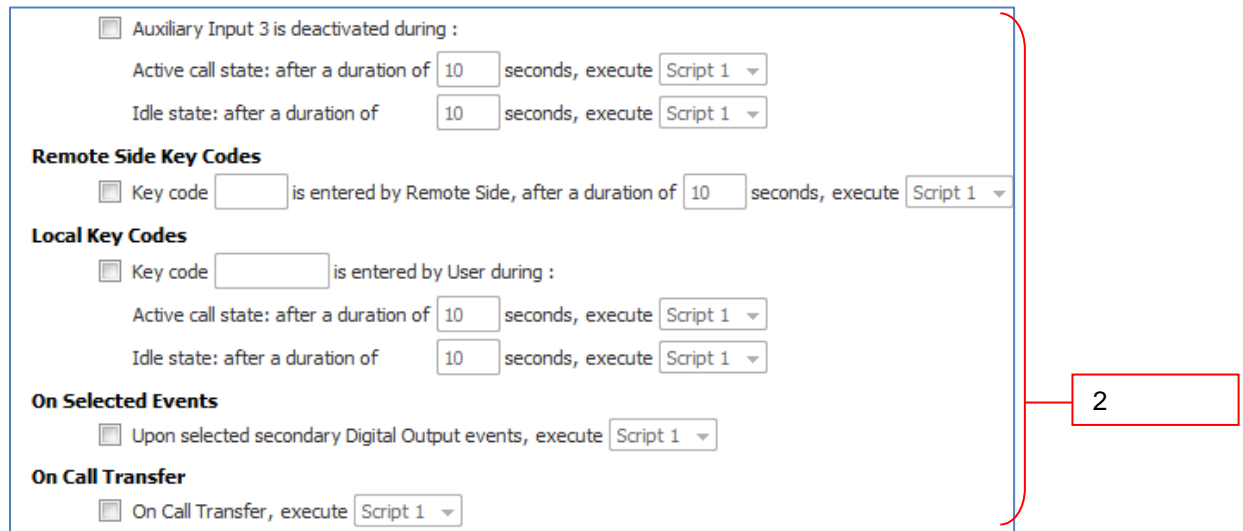


Figure 19 Digital Outputs - Events

To configure the auxiliary outputs:

1. Configure various events that should activate a Digital Output.

i. **Button Events**

The button events are available only when the Buttons are programmed for Autodial mode (Refer to 4.6.1 Buttons: for **Autodial**).

- Configure Primary Digital Output activation on Button #1 and/or Button #2 when a call is **INITIATED, ANSWERED, or TERMINATED**.

ii. **Auxiliary Input Events**

- Configure the Primary Digital Output activation on activation/deactivation of Auxiliary Input #1, Auxiliary Input #2, and Auxiliary Input #3.

iii. **Remote Side Key Codes**

- Configure the Key Code sequence to be used by the remote side to activate the corresponding Digital Output.

iv. **Local Key Codes**

- Configure the Key Code sequence to be used by the Local User to activate the corresponding Digital Output<sup>11</sup>.

2. Configure various events on which the Secondary Digital Output(s) needs to be deactivated. Secondary Digital Outputs are commonly used to send a deactivation script.

i. **Button Events**

The button events are available only when the Buttons are programmed for Autodial mode (Refer to 4.6.1 Buttons: for **Autodial**).

- Configure the Secondary Digital Output activation on Button #1 and/or Button #2 when a call is **INITIATED, ANSWERED, or TERMINATED**, after the defined duration of time has elapsed.

<sup>11</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.  
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ii. **Auxiliary Input Events**

- Configure the Secondary Digital Output activation on activation/deactivation of Auxiliary Input #1, Auxiliary Input #2, and Auxiliary Input #3, during an Active call or while Idle.

iii. **Remote Side Key Codes**

- Configure the Key Code sequence to be used by the remote side to for activation of the corresponding Secondary Digital Output<sup>12</sup>, after the defined duration of time has elapsed.

iv. **Local Key Codes**

- Configure the Key Code sequence to be used by the Local User for activation of the corresponding Secondary Digital Output, during and Active call or while Idle.

v. **On Selected Events**

- Configure the Secondary Digital Output activation on selected events as specified above.

vi. **On Call Transfer**

- Configure the Secondary Digital Output activation on call transfer.

3. Click Apply to save settings.

---

**Note** You cannot use the same event for activation and deactivation of the same Digital Output.

---

---

<sup>12</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.  
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**4.8. Voice Messages**

**4.8.1. Played to the User**

This configuration page allows the administrator to configure the voice message to be played to the local user on hands-free speaker or handset depending on the device mode.

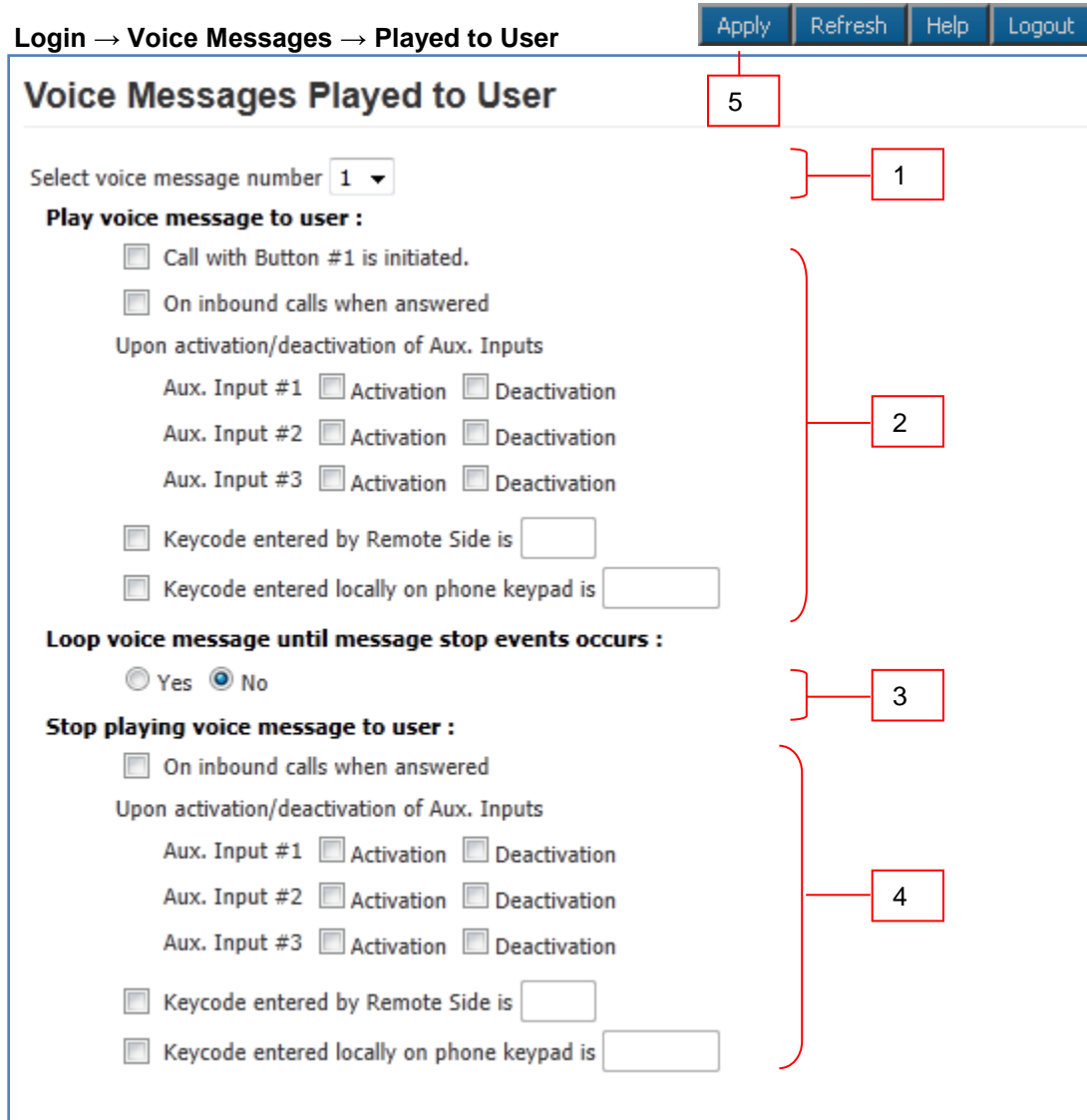


Figure 20 Voice Messages Played to User

To configure the voice messages played to user:

1. Use the drop-down menu to select the voice message to be configured.
2. The voice message is played to the user on the selection of one of the following events:
  - i. Select **On inbound calls when answered** to play the voice message to the user when the phone answers an inbound call.
  - ii. Select Auxiliary Input Activation/Deactivation events to play the voice message to the user on the activation/deactivation of an Auxiliary Input device

- iii. Select and enter the Key Code sequence<sup>13</sup> entered by the remote side to play the voice message to the User when the corresponding key sequence is entered by the remote side during an active call.
  - iv. Select and enter the Key Code sequence<sup>14</sup> entered locally on the phone keypad to play the voice message to the User when the corresponding key sequence is entered by the local User during idle/active call state.
  - v. Select the Button #3 and Button #4 events to play the voice message when the Button #3 or Button #4 is pressed by the local User on the phone. The Button #3 and Button #4 events are available only for the buttons programmed as “Play Voice Message” (Refer to 4.6.1 Buttons).
3. Select Looping – Yes/No to play the voice message in a loop until the “stop message” event occurs.
  4. The selected voice message stops playing on the selection of one of the following events:
    - i. Select **On inbound calls when answered** to stop playing the current voice message when the phone answers an inbound call.
    - ii. Select Auxiliary Input Activation/Deactivation events to stop playing the current voice message on activation/deactivation of an Auxiliary Input Device.
    - iii. Select and enter the Key Code sequence entered by the remote side to stop playing the current voice message when the corresponding key sequence is entered by the remote side during an active call.
    - iv. Select and enter the Key Code sequence entered locally on the phone keypad to stop playing the current voice message when the corresponding key sequence is entered by the local User during idle/active call state.
    - v. Select the Button #3 and Button #4 events to stop playing the current voice message when the Button #3 or Button #4 is pressed by local User on the phone. The Button #3 and Button #4 events are available only for buttons programmed as “Play Voice Message” (Refer to 4.6.1 Buttons).
  5. Click **Apply** to save settings.

---

**Note:** You cannot use the same event to activate and deactivate the same voice message.

---

<sup>13</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

<sup>14</sup> If DTMF Authentication is enabled (see section 4.8) the Guard Access Code will need to be entered before any other code may be used.

4.8.2. Played to Remote Side

**Note** Not all VoIP topologies support this feature. Contact Talkaphone for further information.

This configuration page allows the administrator to configure the voice message that needs to be played to the remote side.

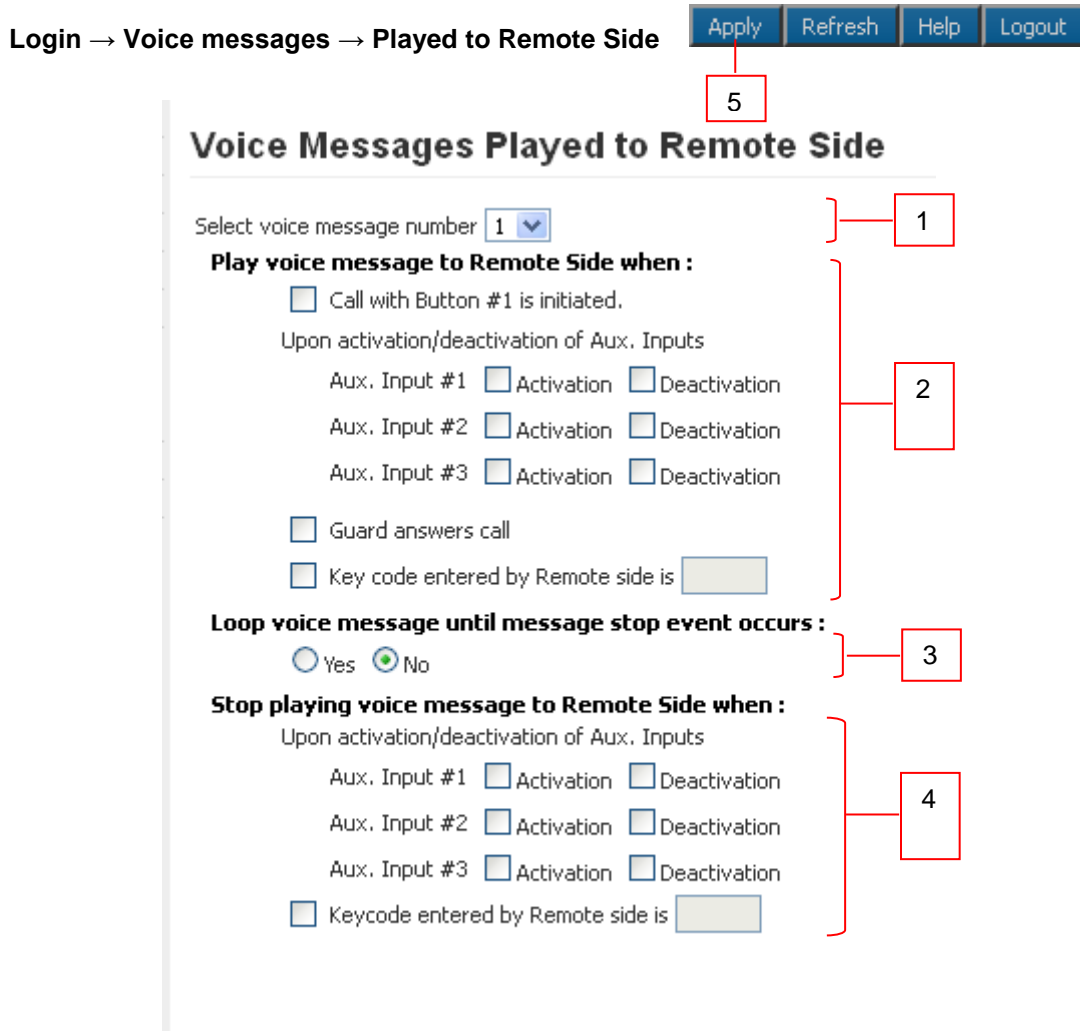


Figure 21 Voice Messages Played to Remote Side

To configure the voice messages played to remote side:

1. Use the drop-down menu to select the voice message to be configured.
2. The voice message is played to the remote side on selection of one of the following events:
  - i. Select **Call with Button #1 is initiated** to play the voice message to remote side when the Button #1 is pressed.
  - ii. Select **Auxiliary Input Activation/Deactivation events** to play the voice message to the Remote side on the activation/deactivation of an Auxiliary Input device

- iii. Select **Guard Answers Call** to play the voice message to remote side when the remote phone answers the call.
  - iv. Select and enter the **Key Code sequence** entered by remote side to play the voice message to the remote side when the corresponding key sequence is entered by the remote side during an active call.
3. Select Looping – **Yes/No** to loop the voice message until the “stop message” event occurs.
  4. The selected voice message stops playing on selection of the following events:
    - i. Select **Auxiliary Input Activation/Deactivation** events to stop playing the current voice message on activation/deactivation of an Auxiliary Input Device.
    - ii. Select and enter the **Key code sequence entered by Remote Side** to stop playing the voice message to remote side when the corresponding key sequence is entered by the remote side during an active call.
  5. Click **Apply** to save settings.

---

**Note** You cannot use the same event to activate and deactivate the same voice message.

---

6.

#### 4.9. Self Diagnostics and Reporting

This configuration page allows the administrator to schedule Self Diagnostics and report the results via a call or an e-mail to the selected users on a daily or a weekly basis.

##### Login → Self Diagnostics and Reporting

Apply Refresh Help Logout

### Self Diagnostics & Reporting

**Schedule Self Diagnostics**

Select Days of Week:

<input type="checkbox"/> Sunday	<input type="checkbox"/> Thursday
<input type="checkbox"/> Monday	<input type="checkbox"/> Friday
<input type="checkbox"/> Tuesday	<input type="checkbox"/> Saturday
<input type="checkbox"/> Wednesday	<input type="checkbox"/> Daily

Specify Time: Hour  Minutes

**Auxiliary Output to be Activated for the Wide-Area Emergency Broadcast Systems (WEBS@) Diagnostic**

Select Auxiliary Output

**Self Diagnostics Results to be Reported**

	Activate Auxiliary Output			Report via Call	Report via Email
	1	2	3		
Button/Switch Integrity	<input checked="" type="radio"/> On Failure <input type="radio"/> Always	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Speaker & Microphone	<input checked="" type="radio"/> On Failure <input type="radio"/> Always	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
WEBS@	<input checked="" type="radio"/> On Failure <input type="radio"/> Always	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

**Configure Speaker/Microphone/Line level Output Gain**

Speaker Gain

Line Level Output Gain

Microphone Gain

**Reporting Self Diagnostics Result**

Enter Number to Call

Enter Email ID

**Deactivation of Auxiliary Outputs**

Upon next scheduled self-diagnostic

Custom

Duration  seconds

7

1

2

3

4

5

6

1. To **Schedule Self Diagnostics** select the Days of the week and **Specify Time** of the day when the self-diagnostics will be performed on the Talkaphone VOIP-600.
2. Select the **Auxiliary Output to be Activated** for the Wide-Area Emergency Broadcast Systems (WEBS@) Diagnostic. Select the Auxiliary Output which connects to the Amplifier of the WEBS@ system.

3. Select the devices for which **Diagnostics Results is to be reported**
  - i. **Button/Switch Integrity:** Select a report to be sent either on **Failure** of the Button or the Switch or send **Always**. Select Activate an Auxiliary Output, Report via Call and/or Report via E-mail as a notification mode for the report.
  - ii. **Speaker & Microphone:** Select a report to be sent either on **Failure** of the Speaker or the Microphone or send **Always**. Select Activate an Auxiliary Output, Report via Call and/or Report via E-mail as a notification mode for the report.
  - iii. **WEBS®:** Select a report to be sent either on **Failure** of the WEBS® or send **Always**. Select Activate an Auxiliary Output, Report via Call and/or Report via E-mail as a notification mode for the report.
  
4. Configure the **Speaker/Microphone/Line-level gain**. Default Speaker gain value is '2', Line Level Output gain value is '2' and Microphone gain value is '12' which provide accurate results for the Self Diagnostics.
  
5. **Enter a number to call** and/or **Enter an E-mail ID** to send a report with the results of the Self Diagnostics.
  
6. Select the event to **De-activate the Auxiliary Output**. De-activate the Auxiliary Output upon next scheduled Self-Diagnostic OR De-activate the Auxiliary Output after the programmed custom duration seconds.
  
7. Click **Apply** to save settings.

#### 4.10. Authentication

The Authentication page allows the administrator to change the user credentials used to access the phone through the Web interface, set the guard access code, and enable/disable the DTMF authentication requirement for specific DTMF operation codes.

The screenshot shows the 'Authentication' configuration page. At the top, there are buttons for 'Apply', 'Refresh', 'Help', and 'Logout'. Below these is the 'Authentication' title. The page is divided into several sections:

- Authentication for HTTP Management Interface :** This section contains three input fields: 'Username' (with 'admin' entered), 'Password', and 'Confirm Password'. A red bracket on the right groups these three fields with the number '1'.
- DTMF Authentication Code :** This section contains a 'Guard Access Code \*4\*' input field. A red bracket on the right groups this field with the number '2'.
- Require DTMF Authentication Code for the following :** This section contains four checkboxes: 'Volume adjustments during call' (unchecked), 'Speaker and Microphone control during call' (checked), 'Voice Messages' (checked), and 'Aux. Output Activation/Deactivation' (checked). There is also a checkbox for 'Help on the Way™ LED Activation/Deactivation' (unchecked). A red bracket on the right groups these checkboxes with the number '3'.
- Check 'SSH' and click on 'Apply' to enable SSH :** This section contains a checkbox for 'SSH' (unchecked). A red bracket on the right groups this checkbox with the number '4'.
- Authentication for SSH Interface :** This section contains three input fields: 'Username' (with 'root' entered), 'Password', and 'Confirm Password'. A red bracket on the right groups these three fields with the number '5'.

At the top of the page, a red box with the number '6' is positioned above the 'Authentication' title.

Figure 22 Authentication

To configure the authentication details:

1. Enter the user credentials to access the Web interface:

- **Username**<sup>15</sup> accepts alphanumeric and special characters (e.g., underscore “\_” and period “.”).
- **Password**<sup>16</sup> accepts alphanumeric values and special characters (except “&”).
- **Confirm password** requires you to type the password again for reconfirmation.

2. Enter the **Guard Access Code**. It is used for authentication when the remote side phone supports the sending of DTMF. The length of the Guard Access Code can be up to eight characters.

<sup>15</sup> The maximum number of characters in username is 30.

<sup>16</sup> The maximum number of characters in password is 30.

3. Check/uncheck the check boxes to enable/disable the DTMF authentication code requirement for specific DTMF operation codes.
4. Check/uncheck the check box to Enable **SSH** access to the VOIP-600 phone.
5. Enter **Username** and **Password** to configure/modify Authentication for SSH user.
6. Click **Apply** to save settings.



**4.11. Maintaining the Phone**

**4.11.1. Device State**

The Device State page displays the current state of the devices that are attached to the phone. If the status indicator is green the device is active, otherwise grey indicates that the device is inactive.

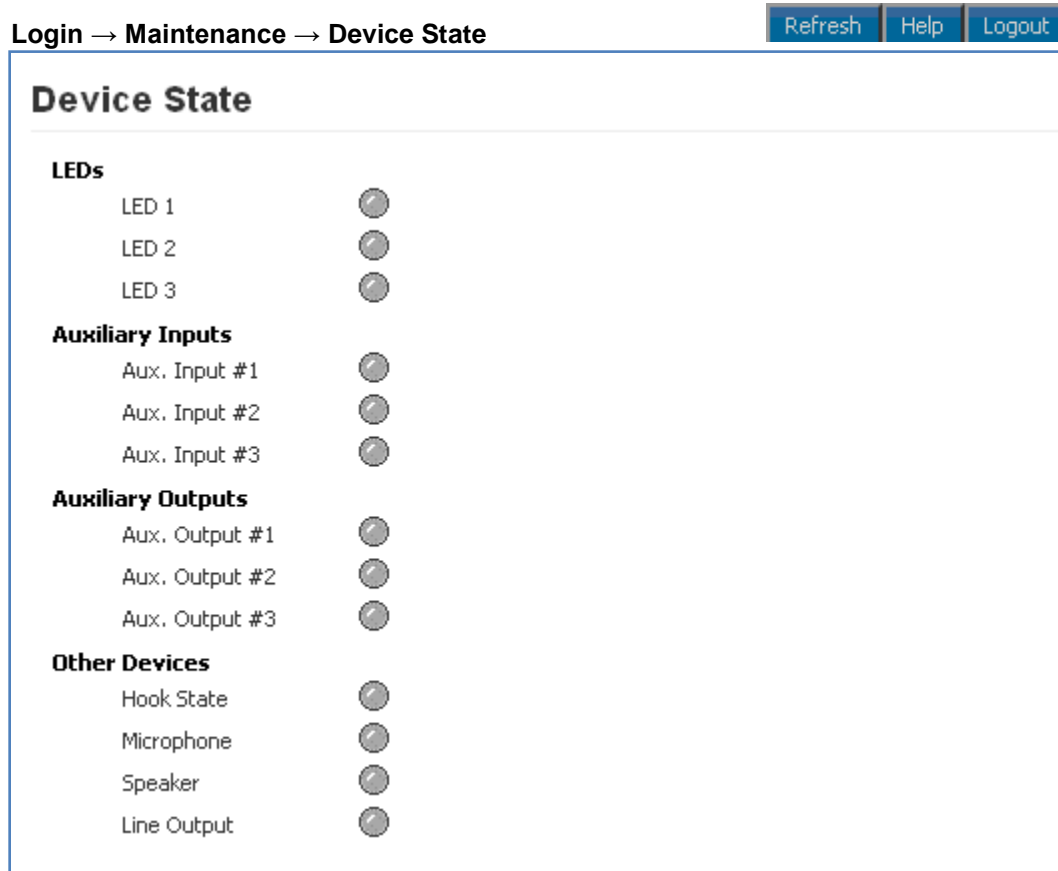


Figure 23 Device state

**Note** The contents of this page will vary depending on the device mode.

4.11.2. Statistics

The Statistics page allows you to manage and maintain the details of the phone calls, current state of the phone, call statistics, call details, and auxiliary details. It also allows you to clear the statistics and provides a link to download the statistics to a text file.

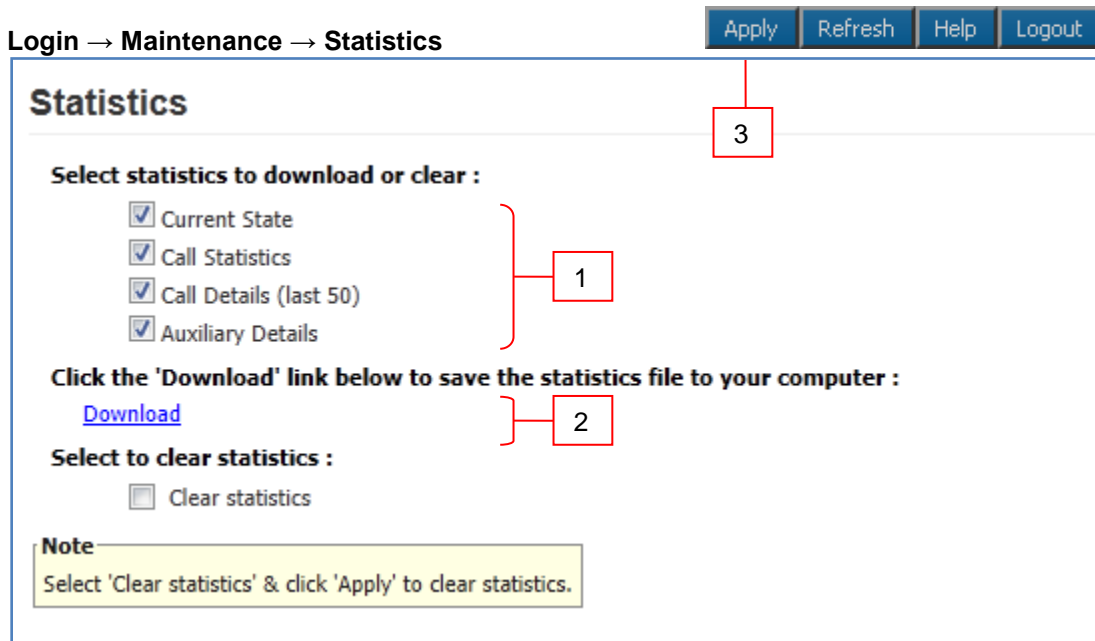


Figure 24 Statistics

To download the statistics file:

1. Select or unselect a specific category of statistics. By default, all options are selected.
2. Click the Download link to download the statistics file to the local computer. A prompt should appear requesting to “save or open” the Statistics File. Click the **Save** option to save the file on the local PC.
3. Click **Apply** to save settings.

**Note** To clear statistics, check the checkbox next to “Clear statistics” and click **Apply**. Executing the “Clear statistics” option will erase all of the selected statistics from the phone.

**Description of Fields in the Statistics File:**

**Call Statistic Fields:**

Field	Description
<b>Outgoing Calls</b>	
Total outgoing call attempts	Count of outgoing call attempts made from Buttons, Keypad, and Auxiliary Input activations.
Total unanswered outgoing calls	Count of unsuccessful calls unanswered by the remote side.
Total failed outgoing calls	Count of failed calls due to events such as remote side is busy, network issues, or unreachable destination.
Total outgoing call duration	Total duration of the successful outgoing calls (hh:mm:ss)

Field	Description
	between call answer and call termination events.
<b>Incoming Calls</b>	
Total incoming call requests	Total number of incoming call request from the remote side.
Total missed calls	Counts the missed calls (calls that are not answered).
Total failed calls	Counts the total number of failed calls (e.g., rejecting an incoming call when an Emergency Call is in progress).
Total incoming call duration	Total duration (hh:mm:ss) of successful incoming calls between call answer and call termination events.
<b>Incoming Pages</b>	
Total incoming page requests	Count of total number of incoming page requests from a WEBS Contact <sup>®</sup> Server.
Total unsuccessful pages	Count of unsuccessful page requests to the phone (e.g., rejecting a page request when an Emergency Call is in progress and has a higher operation mode priority over paging).
Total voice page time	Total duration (hh:mm:ss) of successful pages.

**Call Logs Fields:**

Field	Description
Date	mm:dd:yyyy hh:mm:ss The starting time of a call event or a page event.
Type	Indicates whether the event was an incoming call, outgoing call, or a page.
Status	Displays the status of the call/page (i.e. whether the call/page was completed successfully or not).
Originator	The interface on the phone which initiated the call. The valid entries are: Button #1/Button #2/Button #3/Button #4 AuxIn #1/AuxIn #2/AuxIn #3 Keypad Remote For paging, the Originator is always Remote.
Source URL	URL of the source of the call (maximum of 30 characters).
Destination URL	URL of the destination of the call (maximum of 30 characters).
Duration	hh:mm:ss This indicates how long (in seconds) the call event or page event lasted. This field is only valid for successful calls/pages.
Voice Codec Used	The voice codec being used for a given call. The valid entries are: G.723, G.729, G.711 A law, and G.711 U law

Field	Description
	This field is only valid for successful calls/pages.
Mode	The VoIP protocol used for a particular call (SIP).
Fail Code	The reason code for the failure of a particular call/page. This field is only valid for unsuccessful calls.

**Aux Logs Fields:**

Field	Description
Date	mm:dd:yyyy hh:mm:ss The date and time for an Auxiliary activation/deactivation.
Aux Type	Aux Input 1/2/3 or Aux Output 1/2/3
Aux State	Displays the status of an Aux Input or Aux Output (Activated/Deactivated).

4.11.3. Logging

Log messages generated by the phone can be captured for review.

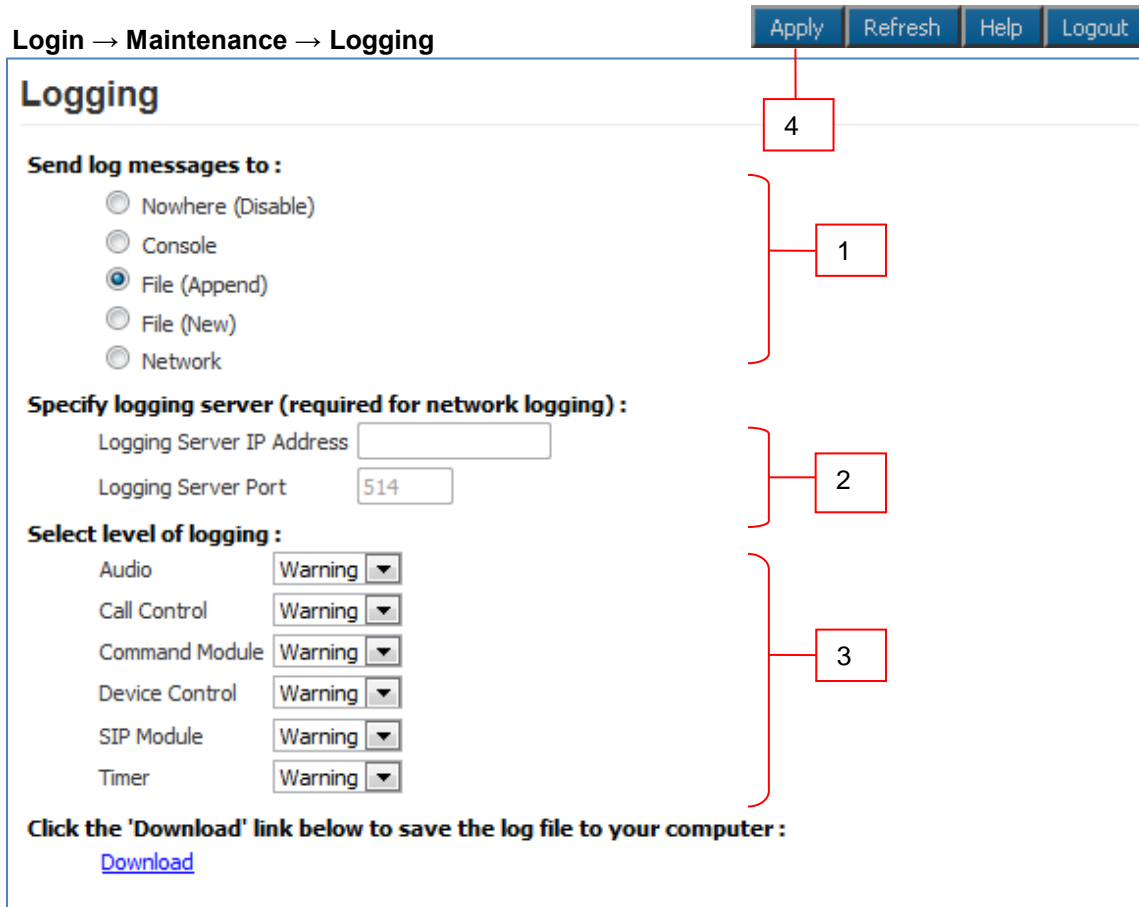


Figure 25 Logging

To configure logging over the network:

1. Select the **Network** option. This option enables **Logging Server** fields.
2. Enter the **Logging Server IP Address** and the **Logging Server Port**.
3. Select the log levels for different modules from the drop-down menu.
4. Click **Apply** to save the settings.

**Note** To disable logging, select **Nowhere (Disable)**

**Note** To Log to Console, connect to the phone as described in section 3.3 and select 'Send Log messages to Console'

**Note** To download and save the log file on the computer, click the Download link.

**Note** A maximum of 500 KB of logging data can be collected. On exceeding this limit, the older 250 KB of logging data are purged from the log file.

4.11.4. Date and Time

The Date and Time page allows you to set the date and time of the phone.

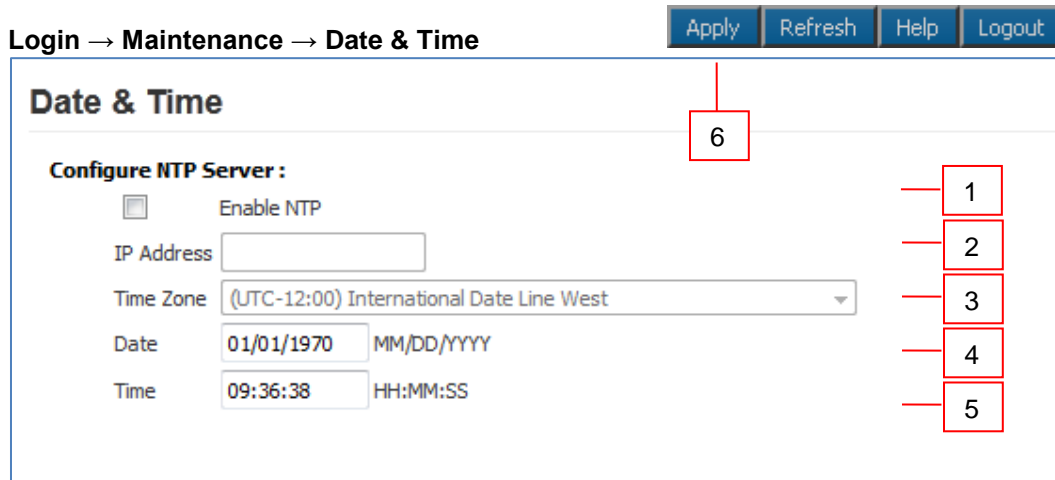


Figure 26 Date and Time Setting

To set the date and time:

1. It is highly recommended to **turn OFF NTP** for a stable operation of the VOIP-600
2. If manually entering the date and time, enter the **Date** in MM/DD/YYYY format.
3. If manually entering the date and time, enter the **Time** in HH:MM:SS 24-hour format.
4. Click **Apply** to save settings.

4.11.5. Ping and Traceroute

The Ping and Traceroute page is used to perform connectivity tests. The Ping command can be used to determine whether a destination is reachable from the phone or not. Traceroute helps in determining intermediate hops to the destination IP address.

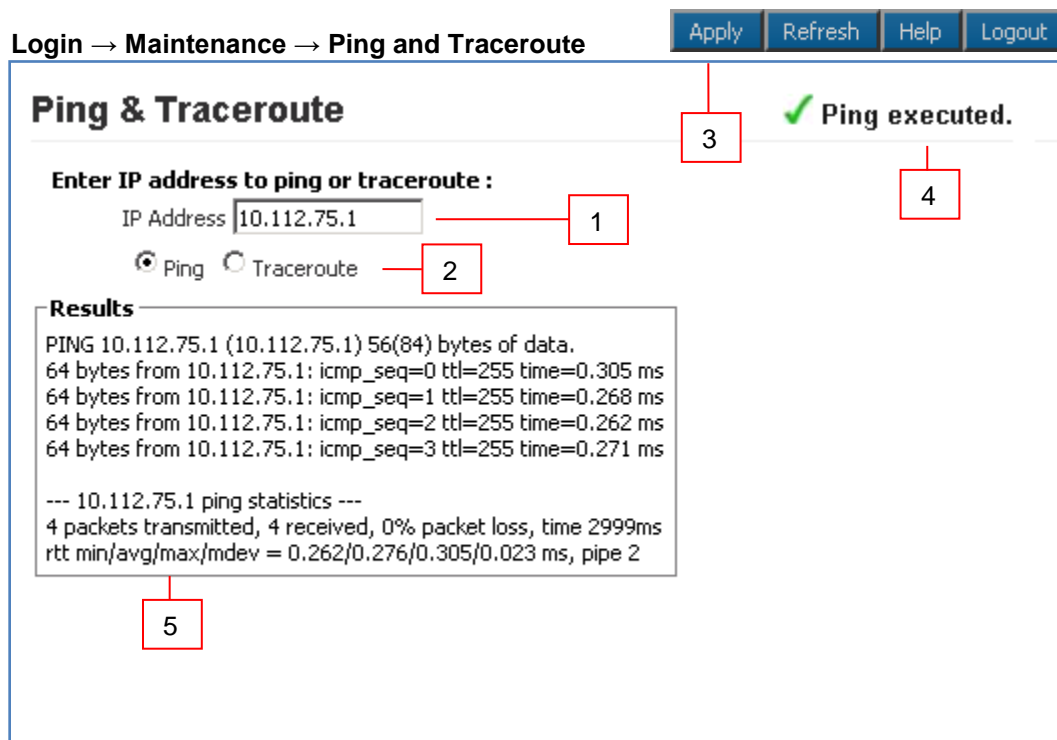


Figure 27 Ping and Traceroute

To ping or traceroute:

1. Enter an **IP Address**.
2. Select **Ping** or **Traceroute**.
3. Click **Apply**.
4. The status bar indicates the command executed.
5. The **Results** section displays the results of the executed command.

4.11.6. Firmware Upgrade

The Firmware Upgrade page is used to update phone to new firmware releases.

Apply Refresh Help Logout

**Login → Maintenance → Firmware Upgrade**

### Firmware Upgrade

**Select protocol to update system:**

FTP  
 HTTP

**Specify firmware/configuration update server :**

Update Server IP Address

Username

Password

Save Username and Password

**Specify filename with complete path :**

Filename

**Check 'Update' and click on 'Apply' to begin the update process :**

Update

**Note**

The update process must not be interrupted.

**Click the 'Download' link below to save the configuration file to your computer :**

[Download](#)

Figure 28 Firmware Upgrade

To upgrade the firmware:

1. Select 'FTP'. Firmware upgrade through HTTP is not supported as of yet.
2. Enter the **FTP Server IP Address**, **Username**<sup>17</sup>, and **Password**<sup>18</sup> for authentication. If the username and password is preferred to be retained for a future upgrade, select the **Save Username and Password** check box.
3. In the **Filename** box, enter the filename with its complete path. If the file is saved in the home directory of the ftp server, then enter ONLY the filename and NOT the complete filepath. The filename is generally in the format of (EVP\_A\_B\_C\_D). If the firmware upgrade file is **not** in the home directory of the ftp server, then this filename needs to be prefixed with directory relative to the FTP root directory. For example, /ftp/EVP\_images/EVP\_1\_0\_0\_0). Consult the Administrator for FTP settings.

<sup>17</sup> The maximum number of characters in username is 30.

<sup>18</sup> The maximum number of characters in password is 30.



4. Select the **Upgrade** check box.
6. Click **Apply**. One of the following progress messages is displayed:
  - **Firmware upgrade initiated. Do not reboot...:** The firmware upgrade has been initiated; the firmware upgrade request is processed only when phone is in an idle state. For example, a call is in progress and administrator requests for a firmware upgrade through the Web GUI. The phone checks for an active call every five seconds and starts the process once the active call is terminated.

The following are considered to be idle phone states: no active call, no dialing, no voice message is being played, and the Auxiliary Output is deactivated.

- **Firmware upgrade in progress. Do not reboot...:** The firmware upgrade is in progress. The process generally takes around 1-2 min to complete.
- **Firmware upgrade done. Going to reboot...:** The firmware upgrade is complete and a reboot has been initiated. After approximately one minute, the phone will be accessible once again.

#### 4.11.7. SMTP Server Configuration

The SMTP Server Configuration page is used to configure the mail settings for email notifications.

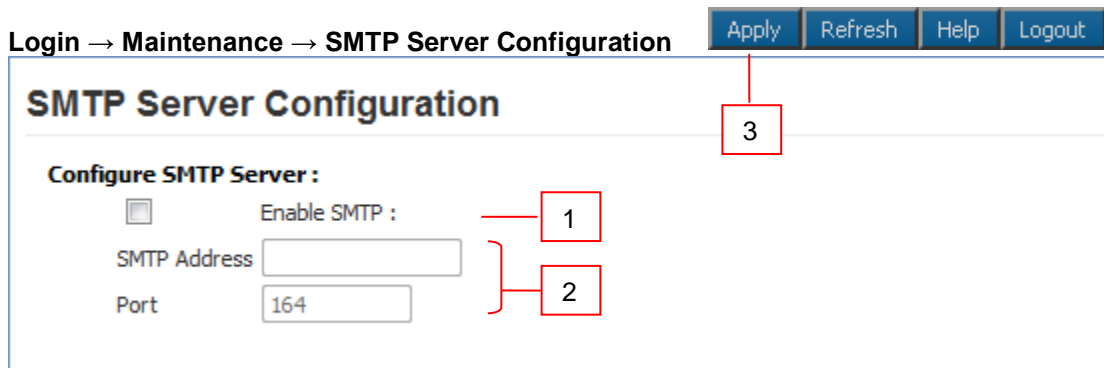


Figure 29 SMTP Server Configuration

To setup SMTP:

1. Check the **Enable SMTP** check box.
2. Enter the **SMTP Address** and **Port** where emails should be sent.
3. Click **Apply** to save settings.

**Note:** VOIP-600 does not support 'Authentication with the mail server'. Hence, if it is desired that an e-mail be sent out via such a Mail Server which requires authentication, a 'Mail Relay' will have to be configured to achieve this and it's IP address should be put in the 'SMTP Address'.

4.11.8. Email Notification Profiles

The Email Notification Profiles page is used to create email notification profiles and select the events that will trigger an email message being sent.

Login → Maintenance → Email Notification Profiles

Apply Refresh Help Logout

### Email Notification Profiles

**Select Profile :**  
Select voice Notification Profile number  1

**Select Notification Events :**

<input type="checkbox"/> Emergency Call	<input type="checkbox"/> Aux Input 1 Activation
<input type="checkbox"/> Information Call	<input type="checkbox"/> Aux Input 2 Activation
<input type="checkbox"/> System Restore - Ethernet up	<input type="checkbox"/> Aux Input 3 Activation
<input type="checkbox"/> System Restore - Firmware Upgrade Fail	<input type="checkbox"/> Aux Input 1 Deactivation
<input type="checkbox"/> SIP Registration	<input type="checkbox"/> Aux Input 2 Deactivation
<input type="checkbox"/> SIP Deregistration	<input type="checkbox"/> Aux Input 3 Deactivation
<input type="checkbox"/> Aux Output 1 Activation	<input type="checkbox"/> Aux Output 2 Activation
	<input type="checkbox"/> Aux Output 3 Activation

**Enter the Email Addresses Associating with this profile :**

1

2

3

4

5

Figure 30 Email Notification Profiles

To setup an Email Notification Profile:

1. Check the **Notification Profile Number** you wish to modify. A total of 3 profiles are available.
2. Check the box for each **Notification Event** that should trigger an email notification being sent.
3. Enter the **Email Addresses** that the email notification should be sent to.
4. Click **Apply** to save settings.

4.11.9. Reset to Defaults

The Reset to Defaults page allows the phone configuration parameters to be set to their default values. It also provides an option to retain the IP address and/or the username and password.

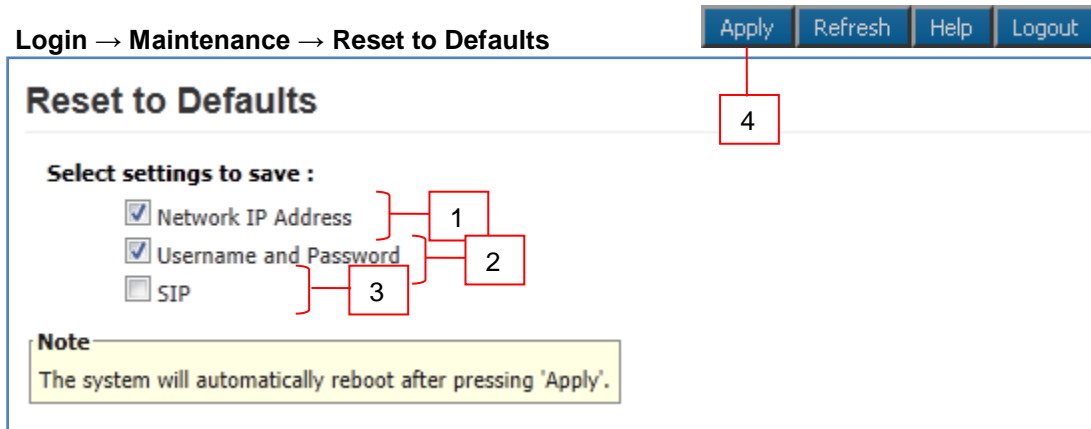


Figure 31 Reset to Defaults

To reset to defaults:

1. Select **Network IP Address**, if the IP address of the phone needs to be retained. If this option is not selected the IP address of the phone will default to 192.168.1.10.
2. Select **Username and Password** if the Web Interface credentials need to be retained. If the option is not selected the username will default to admin and the password will default to admin@123.
3. Select **SIP** if the SIP settings need to be retained. If this option is not selected the SIP Settings will be deleted.
4. Click **Apply**. The phone automatically reboots with the default settings.

4.11.10. Reboot

The reboot page allows the device to be rebooted via the GUI.

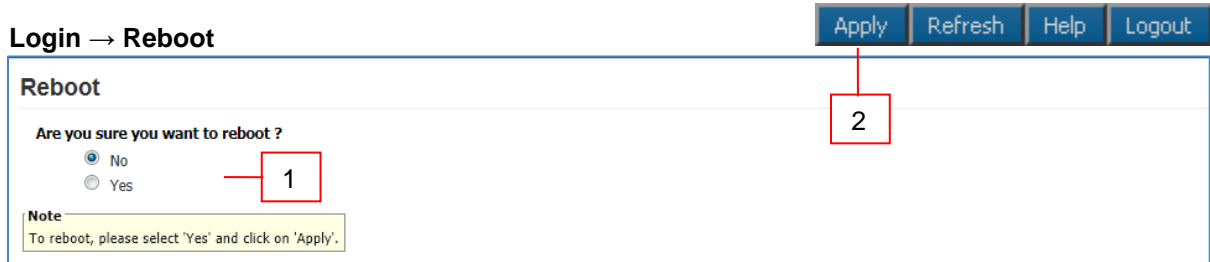


Figure 32 Reboot

To reboot the device:

1. Select **Yes**.
2. Click **Apply**. The phone automatically reboots with the default settings.

## 5. Operations

The phone takes approximately one minute to boot. While the first and second LED's are flash, the unit is still booting (Refer to Figure 1 Front View). After all LED's clear, wait for an additional twenty seconds before initiating a call.

The phone needs to be programmed in accordance with the instructions provided in Chapter 4 (Refer to 4 Using the Web GUI).

### 5.1. Calling Emergency/Information Numbers

In case of an emergency, the phone can dial a list of numbers (Refer to 4.5.1 Number Lists) in round robin fashion. Depending on the device mode and programming, a call is automatically initiated when:

- Push button is pressed
- Auxiliary Input is activated
- Handset goes off-hook (OBH device mode only)

If the first number does not answer within a specified time (Refer to 4.5.4 Call Parameters) the phone dials the next number from the number list. It continues to dial in round robin fashion until the call is answered, or the call conversation timer expires (Refer to 4.5.4 Call Parameters: Call Conversation Timer).

### 5.2. Answering the Emergency/Information Call

One of the following scenarios occurs upon answering the emergency call on the remote side:

- The VOIP-600 Series Phone may transmit its location by playing a recorded message (Refer to 4.7.2 Played to Remote Side). If no such feature is enabled or when the voice message completes playing, the caller may talk and listen hands-free<sup>19</sup> without using any controls.
- The VOIP-600 Series Phone may transmit a voice message and request that the remote side user to press a specific key on the keypad of the remote side phone (Refer to 4.5.4 Call Parameters: answer a call) to answer this call. Providing the incorrect key sequence within three attempts will terminate the call. Once the call is terminated, a new call is made to the next number on the list. On providing the correct key sequence, the VOIP-600 Series Phone may transmit its location by playing a recorded message (Refer to 4.7.2 Played to Remote Side). If no such feature is enabled, the caller may talk and listen hands-free<sup>20</sup> without using any controls.

### 5.3. Terminating Calls

Round robin calls can be terminated only from the remote side by going on-hook or by pressing a specific call termination key on the remote keypad (Refer to 4.5.4 Call Parameters). Calls placed from the keypad can be terminated locally using button(s) programmed as a hook switch (Refer to 4.6.1 Buttons) or from the remote side by going on-hook/pressing a specific call termination key.

For phones with just a handset and no speaker, calls (both round robin and keypad calls) can be terminated locally by going on-hook.

---

<sup>19</sup> Depends on device mode

<sup>20</sup> Depends on device mode

#### **5.4. Activating/Deactivating Auxiliary Outputs**

To provide additional visible or audible alerts upon an event, the phone is equipped with 3 Auxiliary Outputs. The Auxiliary Outputs can be used to connect devices such as a siren, a strobe light, a PTZ camera, etc. These devices can then be activated and deactivated locally or remotely (Refer to 4.6.6 Auxiliary Outputs).

#### **5.5. Activating/Deactivating the “Help on the Way” LED**

The phone is equipped with 3 LEDs; each of the LEDs has a specific function assigned to it. The third LED is used for indicating “Help on the Way” and is activated/deactivated through DTMF operation codes on the remote side (Refer to 4.6.4 LEDs).

#### **5.6. Activating/Deactivating Speaker and Microphone**

During a call, the speaker (both Hands-free and Handset) and microphone (both Hands-free and Handset) can be enabled/disabled using DTMF operation codes \*5\*/\*6\* & \*7\*/\*8\* respectively.

Note: The settings made above are temporary, which are valid for current call only. Once the call is terminated, the settings will reset to the values defined in the Web GUI (Refer to 4.5.3 Audio Settings: Enable or disable the Speaker and/or Microphone).

#### **5.7. Adjusting Volume**

During a call, the speaker gain and the microphone gain can be adjusted remotely using DTMF operation codes \*32\*level\* and \*31\*level\* (where level ranges from 1-20), respectively.

During a call, the User can also locally adjust the speaker volume using Button #3 and Button #4 (Refer to 4.6.1 Buttons). This option is only available for phones with four buttons.

Note: The volume adjustments made above are temporary and are valid for a single active call instance. Once the call is terminated, the settings will reset to the values defined in the Web GUI (Refer to 4.5.3 Audio Settings: Speaker Gain and the Microphone Gain).

#### **5.8. Playing Voice Messages**

This feature is used to play pre-recorded voice messages on the phone’s built-in speaker/handset or to the remote side upon occurrence of certain events (e.g., Call Initiation events, local/remote key press events). For configuration details, refer to Section 4.7.1 Played to the User and Section 4.7.2 Played to Remote Side.

### 5.9. Recording Voice Messages

The VOIP-600 Series Phone has the capability of recording up to six different voice messages (each up to one minute in length).

To record a voice message on the phone from the remote side:

1. Call the VOIP-600 Series Phone.
2. Once the call is answered, enter the DTMF authentication code. The default DTMF authentication code is \*4\*\*.
3. To initiate voice message recording, press \*91\*<message num>\* where message num is from 0-5. The voice message programmed for message num 0 is used for the “Key Press to Answer” feature.
4. After the single beep, begin recording the voice message.
5. Press any key to stop the voice recording.

---

**Note** The maximum voice message length is one minute. If no key is pressed within one minute or the call is disconnected, the voice message recording stops automatically.

---

### 5.10. Previewing Voice Messages

To preview your recorded voice messages from the remote side:

1. Call the VOIP-600 Series Phone.
2. Once the call is answered, enter the DTMF authentication code. The default DTMF authentication code is \*4\*\*.
3. To initiate voice message previewing, press \*92\*<message num>\* where message num is from 0-5. The voice message programmed for message num 0 is used for the “Key Press to Answer” feature.
4. After the single beep, the voice message will begin playing.
5. Press any key to stop the preview.

### 5.11. Feedback Tones

Feedback tones/beeps are used to indicate the status of the requested operation from the local keypad or from the remote side.

1. One beep is played after successful translation of local/remote key codes. For example, to activate Auxiliary Output #1 the local key code is 123. When the User presses 123, the phone plays a single beep on the built-in speaker or handset before activating Auxiliary Output #1. Similarly, when the remote side enters the correct code the remote user hears a single beep before the phone performs the requested operation.
2. Two beeps are played indicating an incorrect key code. For example, to activate an Auxiliary Output #1 the local key code is 123. If the User presses 124 the phone plays two beeps. Similarly, when the remote side enters the incorrect code the remote user hears a double beep indicating an error.
3. Three beeps are played to the remote side in a case where DTMF authentication is required for a particular operation. For example, key code 123 requires DTMF authentication to activate Auxiliary Output #1. After pressing 123 the phone plays three beeps to the remote side indicating DTMF authentication is required in order to perform the requested operation.

### 5.12. Paging Operation

The phone can receive one-way paging messages from the WEBS Contact® server either on the line level output or on the built-in hands-free speaker. The phone must be registered with the WEBS Contact® server in order to receive incoming pages.

The system administrator can prioritize (Refer to 4.5.2 Phone Settings: assign the priority) the paging operation over the phone operations to facilitate high priority paging messages.

### 5.13. Line Level Recording

This feature is used to locally record communication between the calling party and the called party once the call is established. This feature is enabled only when the paging output mode is set to the built-in hands-free speaker. Configuring the paging output to line level output automatically disables this feature.

### 5.14. Silent Monitoring

Silent Monitoring is used to monitor locations as required. This feature is only supported by phones with a built-in hands-free speaker and microphone. To use this feature:

1. Disable the speaker (Refer to 4.5.3 Audio Settings: Enable or disable the Speaker and/or Microphone) and enable the auto answer mode of operation.
2. Place an incoming call to the VOIP-600 Series Phone. The phone will answer the call silently. The remote phone can be used to monitor the nearby situation without any audible or visual indication on the phone.
3. Please be certain to follow all federal, state, and local laws when using this feature.



### 5.15.Rebooting the Phone

The phone can be rebooted using the Reset button (Refer to 2.1.2 Internal View: RESET BUTTON) on the phone. Pressing the Reset button once will reboot the phone. The phone can also be rebooted via the Web GUI (Refer to 4.10 Reboot).

### 5.16.Factory Default Settings

The phone can be reset to factory settings using the Reset button (Refer to 2.1.2 Internal View: RESET BUTTON). Press and hold the Reset button for 10 seconds. The phone will reboot with factory default settings.

---

Note: All data including IP address and username/password will be reset to factory defaults.

---

## Appendix A: Remote Operation Key Codes – Quick Reference

During a call, the following operations can be performed on the phone by dialing a key code sequence from the remote side phone which supports the sending of DTMF<sup>21</sup>. If configured, these operations may require DTMF authentication codes (Refer to 4.8 Authentication: DTMF Authentication Code).

Here is an example of how to use the DTMF operation codes during an active call.

For example, to adjust the volume of speaker from a remote phone, follow these steps:

Authenticate using key sequence \*4\*<authentication code>\*. Successful authentication is indicated by a single beep. This step is required only when DTMF authentication is enabled for volume adjustments (Refer to 4.8 Authentication: enable/disable the DTMF authentication)

Dial the key sequence \*31\*8\*. This sets the volume level of the speaker to 8. Successful operation is indicated by a single beep.

---

Note The DTMF Authentication Code is only required once during a call and is valid until the call is terminated.

---

Operation	Key Code	Function	DTMF Authentication Code Required (if configured)
<b>Enter Configuration Mode</b>	*4*<authentication code>*	Code is used to control access to the device.	
<b>Microphone Gain Adjustment</b>	*31*<Level>* Valid Level range : 1-20	Code is used to control microphone gain levels. This applies to both handset and hands-free models. The code takes effect for the current call only.	Yes
<b>Speaker Gain Adjustment</b>	*32*<Level>* Valid Level range : 1-20	Code is used to control speaker gain levels. This applies to both handset and hands-free models. The code takes effect for the current call only.	Yes
<b>Speaker Enable/Disable</b>	*5* → Enable Speaker *6* → Disable Speaker	Code is used to enable/disable the speaker. This applies to both handset and hands-free models. The code takes effect for the current call only.	Yes
<b>Microphone Enable/Disable</b>	*7* → Enable Microphone *8* → Disable Microphone	Code is used to enable/disable the microphone. This applies to both handset and hands-free models. The code takes effect for the current	Yes

<sup>21</sup> Out-of-band RTP

Operation	Key Code	Function	DTMF Authentication Code Required (if configured)
		call only.	
<b>AUX Output Activation/Deactivation</b>	*11* → Aux 1 Activation *12* → Aux 2 Activation *13* → Aux 3 Activation *21* → Aux 1 Deactivation *22* → Aux 2 Deactivation *23* → Aux 3 Deactivation  These are the default codes and can be changed from Web GUI.	Code is used for activation/deactivation of the Auxiliary Output by the remote side phone during the conversation.	Yes
<b>Voice Message Recording</b>	*91*<Message Number>* Message Number : 0-5 Note: Voice message programmed for Message Number 0 is used for the “Key Press to Answer” feature.	Code is used for recording the message after the beep. Pressing a key stops the recording.	Yes
<b>Voice message previewing</b>	*92*<Message num>* Message num : 0-5 Voice message programmed for Message Number 0 is used for the “Key Press to Answer” feature.	Code is used for previewing a recorded voice message. Pressing a key stops the previewing.	Yes
<b>“Help on the Way” LED Activation</b>	*98* → Enable *97* → Disable  These are the default codes and can be changed from the Web GUI.	Code is used to enable/disable the “Help on the Way” LED.	Yes
<b>Call Termination</b>	# → To disconnect current call  This is the default code and can be changed from the Web GUI.	Single digit key code to be used to disconnect the call.	No

**Appendix B: Troubleshooting & Maintenance**

Problem	Possible Causes & Corrective measures
<b>Unable to access the phone's Web GUI</b>	<ol style="list-style-type: none"> <li>1. The IP address being used in the Web browser is incorrect. Connect the serial console and check the phone's IP address via the 'ifconfig' command.</li> <li>2. IP address of the phone is conflicting with another network device. Reassign a new IP address to the phone.</li> <li>3. The phone's Web server is not responding. Reboot the phone.</li> </ol>
<b>Unable to login to the phone's Web GUI</b>	<ol style="list-style-type: none"> <li>1. The phone's database is corrupt. Reset to factory default settings using the <b>Reset</b> button (Refer to <b>Error! Reference source not found. Error! Reference source not found.</b>).</li> <li>2. IP address of the phone is conflicting with another network device. Reassign a new IP address to the phone.</li> </ol>
<b>"Call Placed" and "Call Received" LED's are constantly blinking</b>	<ol style="list-style-type: none"> <li>1. The phone's Ethernet link is down. Check the Ethernet connection.</li> <li>2. The phone is unable to register to a SIP Registrar. Check SIP settings.</li> <li>3. A firmware upgrade is in progress. Wait until the process is complete.</li> </ol>
<b>Unable to make outgoing calls</b>	<ol style="list-style-type: none"> <li>1. The Phone is not on the network. Check network settings.</li> <li>2. Make sure the entries in the SIP Settings-&gt;Outbound proxy are valid</li> <li>3. The SIP registrar is not reachable. Check SIP settings for Registration Status.</li> </ol>
<b>Unable to receive incoming calls</b>	<ol style="list-style-type: none"> <li>1. The Phone is not on the network. Check network settings.</li> <li>2. The Phone is off-hook. Check the hook switch status.</li> <li>3. The SIP registrar is not reachable. Check SIP settings for Registration Status.</li> </ol>
<b>Not able to recognize the DTMF codes from remote side</b>	<ol style="list-style-type: none"> <li>1. The phone's DTMF settings are not compatible with remote side. Check the DTMF settings of the remote side. It should be RFC 2833 compliant.</li> <li>2. The DTMF digit duration of the phone is not compatible with the remote phone. Check the DTMF settings at the remote side. It should be RFC 2833 compliant.</li> </ol>

## Appendix C: Frequently Asked Questions

1. I am experiencing audio problems on the phone such as echo, distorted sound, or choppiness. How do I fix this?
  - i. Verify whether proper bandwidth is allocated for VoIP traffic. You can utilize various mechanisms (RTP header compression and QoS) on the network.
  - ii. Check the network priority configuration on the Buttons configuration page (Refer to 4.6.1 Buttons: Network Priority).

2. How do I determine the IP address of the phone?

To determine the IP address:

- i. Connect a computer to the phone in question using a serial port and serial console.
- ii. Open HyperTerminal or similar console application on the connected computer. Refer to section 3.2 for console connection settings for VOIP-600.
- iii. At the command prompt, enter the ifconfig command. The IP address of the phone will be displayed.

To determine the IP address through the Web GUI:

- i. Follow the directions under Section 4.4.1 IP Settings.
- ii. The IP Address field will display the IP Address of the phone.

3. The phone is not receiving paging requests from the WEBS Contact® server. How do I fix this?
  - i. Refer to Section 4.5.5 Paging Settings.
  - ii. Check the Registration status with the WEBS Contact® server.
  - iii. If the status is Unregistered then you will need to contact the Administrator of the WEBS Contact® server in order to reregister the phone for paging requests.
  - iv. If the status is registered at <WEBS Contact® server> then refer to Section 4.5.2 Phone Settings.
  - v. Check whether Phone Mode has priority over Paging Mode or Paging Mode has priority over Phone Mode.
  - vi. If Phone Mode has priority over Paging Mode, then check for any phone activity – Is the phone dialing? Is the handset off-hook? In order to receive pages, the phone should be idle.

---

Note It is not recommended to use the deregister option as it deregisters the phone with the WEBS Contact® Server. Phone will stop sending events to WEBS Contact® Server.

---

**Appendix D: VOIP-600 Device Modes**

This section provides detail about the various device modes of the VOIP-600 Series Phone. The device mode is specified under the Home page of the Web GUI (Refer 4.3 Home).

Model Number	Device Mode	Number of Buttons	Handset	Hands-free Speakear/Microphone	Keypad
VOIP-600HNS	0BH	0	Yes	No	No
VOIP-600KHNS	0BKH	0	Yes	No	Yes
VOIP-600H	1BH	1	Yes	No	No
VOIP-600	1BS	1	No	Yes	No
VOIP-600H2	1BSH	1	Yes	Yes	No
VOIP-600KS	1BSK	1	No	Yes	Yes
VOIP-600KSH2	1BSKH	1	Yes	Yes	Yes
VOIP-600KSH	1BKH	1	Yes	No	Yes
VOIP-600DH	2BH	2	Yes	No	No
VOIP-600D	2BS	2	No	Yes	No
VOIP-600DH2	2BSH	2	Yes	Yes	No
VOIP-600K	2BSK	2	No	Yes	Yes
VOIP-600KH2	SBSKH	2	Yes	Yes	Yes
VOIP-600KH	2BKH	2	Yes	No	Yes
VOIP-600-OEM	4BSKH	4	Yes	Yes	Yes
WEBS-CM-2	0BS	0	No	Yes (Speaker only)	No

**Note** 0BS device mode is also referred to as Communications Module.

**Appendix E: Sample Serial Configuration File**

```

+++++
++ This Configuration file has all the default and allowed values for
++ configurable attributes.
++ Admin/Programmer can modify the values as per permissible value/range for
++ defined parameters as described in comments given for each parameter.
++
+++++

```

```

+++++
++ Field name      : VoIPProtocol
++ Description     : Protocol to be used for originating calls
++ Valid values   : VoIPProtocol = 1 for SIP (Default Value)
++                : VoIPProtocol = 2 for H.323
++ Note          : Changing this value will cause the phone to reboot.
+++++

```

VoIPProtocol = 1

```

+++++
++ Field name      : AutoAnswer
++ Description     : Enables/disable auto answer mode of phone operation.
++ Valid values   : AutoAnswer = 0 - Disable
++                : AutoAnswer = 1 - Enable (Default Value)
++ Note          : None
+++++

```

AutoAnswer = 1

```

+++++
++ Field name      : OperationPriority
++ Description     : Operation Priority between Phone and Paging operations.
++ Valid values   : OperationPriority = 1 Phone Mode has priority over Paging
++                : Mode(Default)
++                : OperationPriority = 2 Paging Mode has priority over Phone
++                : Mode
++ Note          : None
+++++

```

OperationPriority = 1

```

+++++
++ Field name      : CallerIDPresentation
++ Description     : Caller ID presentation to the Remote party.
++ Valid values   : CallerIDPresentation = 0 - Disable
++                : CallerIDPresentation = 1 - Enable (Default Value)
++ Note          : None
+++++

```

CallerIDPresentation = 1

```

+++++
++ Field name      : CallerID
++ Description     : Caller ID to be present to Remote Party.

```

```

++ Valid values : 30 chars long, a-z,A-Z,*,0-9,.,- , No spaces allowed.
++           Default value is VOIP-600
++ Note      : This will be sent when CallerIDPresentation = 1
++++

```

CallerID = VOIP-600

```

++++
++ Field name  : AudioCodec
++ Description : Preferred Audio Codec to be used for Outgoing Calls.
++ Valid values : AudioCodec = 1 - G.711 PCM A Law
++           AudioCodec = 2 - G.711 PCM u Law (Default Value)
++           AudioCodec = 3 - G.729a
++           AudioCodec = 4 - G.723.1a
++ Note      : None
++++

```

AudioCodec = 2

```

++++
++ Field name  : AEC
++ Description : Enable/Disable Acoustic Echo Cancellation.
++ Valid values : AEC = 1 - Enable (Default Value)
++           AEC = 0 - Disable
++ Note      : None
++++

```

AEC = 1

```

++++
++ Field name  : VAD_CNG
++ Description : Enable/Disable Voice Activity detection/generation of
++           Silence packets.
++ Valid values : VAD_CNG = 1 - Enable (Default Value)
++           VAD_CNG = 0 - Disable
++ Note      : None
++++

```

VAD\_CNG = 1

```

++++
++ Field name  : AGC
++ Description : Enable/Disable Automatic Gain Control
++ Valid values : AGC = 0 - Disable (Default Value)
++           AGC = 1 - Enable
++ Note      : None
++++

```

AGC = 0

```

++++
++ Field name  : JitterBuffer
++ Description : Size of Jitter buffer (milliseconds)
++ Valid values : 0 (Default Value), 30, 60, 90 , 120
++ Note      : None

```



+++++

JitterBuffer = 30

+++++

++ Field name : LineLevelOutputGain  
 ++ Description : Line level output gain to be used during Recording or  
 ++ Amplifying.  
 ++ Valid values : 1 to 20, Default value is 16  
 ++ Note : None

+++++

LineLevelOutputGain = 16

+++++

++ Field name : Speaker  
 ++ Description : Enable/disable Speaker state, applicable to both Handset  
 ++ and Handsfree  
 ++ Valid values : Speaker = 0 - Disabled  
 ++ Speaker = 1 - Enabled (Default value)  
 ++ Note : None

+++++

Speaker = 1

+++++

++ Field name : SpeakerGain  
 ++ Description : Speaker gain, applicable to both Handset and Handsfree.  
 ++ Valid values : 1 to 20, Default value is 12  
 ++ Note : None

+++++

SpeakerGain = 12

+++++

++ Field name : Mic  
 ++ Description : Enable/disable Mic state, applicable to both Handset  
 ++ and Handsfree  
 ++ Valid values : Mic = 0 - Disabled  
 ++ Mic = 1 - Enabled (Default value)  
 ++ Note : None

+++++

Mic = 1

+++++

++ Field name : MicGain  
 ++ Description : Mic gain, applicable to both Handset and Handsfree.  
 ++ Valid values : 1 to 20, Default value is 12  
 ++ Note : None

+++++

MicGain = 12

+++++

```

++ Field name      : CallProgressTones
++ Description    : Enable/Disable Call Progress Tones (Dial tone,Ringback
++                tone, Ringing tone, Busy Tone etc)
++ Valid values   : CallProgressTone = 0 - Disabled
++                CallProgressTone = 1 - Enabled (Default Value)
++ Note          : None
+++++

```

CallProgressTones = 1

```

+++++
++ Field name      : KeyToAnswer
++ Description    : Key to be pressed by Remote party for answering the call.
++ Valid values   : 0-9, *, #, disable (default value)
++ Note          : Only applicable for Outbound Round Robin calls.
+++++

```

KeyToAnswer = Disable

```

+++++
++ Field name      : KeyToDisconnect
++ Description    : Key to be pressed by Remote party to disconnect the call
++                from remote.
++ Valid values   : 0-9, *, # (default value), disable.
++ Note          : None
+++++

```

KeyToDisconnect = #

```

+++++
++ Field name      : WelcomeTone
++ Description    : Enable/disable playing of Welcome Tone to remote party.
++ Valid values   : WelcomeTone = 0 - To not play
++                WelcomeTone = 1 - To Play (Default Value)
++ Note          : Only played when AutoAnswer = 1.
+++++

```

WelcomeTone = 1

```

+++++
++ Field name      : TimerNetResponse
++ Description    : Network Response Timer
++ Valid values   : 5 - 20 sec, in increment of 1 sec.
++                Default value is 5 secs
++ Note          : Only applicable to Outbound Round Robin Calls
+++++

```

TimerNetResponse = 5

```

+++++
++ Field name      : TimerRinger
++ Description    : Ringer Timer in number of rings.
++ Valid values   : 1 - 12. Default value is 5 rings
++ Note          : Only applicable to Outbound Round Robin Calls
+++++

```

TimerRinger = 5

```

+++++
++ Field name   : TimerHangup
++ Description  : Hangup Timer
++ Valid values : 0.5 - 3.0 seconds, in increment of 0.1 second.
++             : Default value is 0.5 secs
++ Note        : None
+++++

```

TimerHangup = 0.500000

```

+++++
++ Field name   : TimerInterDigitLocal
++ Description  : Local Inter digit timer, used during dialing numbers.
++ Valid values : 5 - 20 seconds, in increment of 1 sec
++             : Default value is 5 secs
++ Note        : None
+++++

```

TimerInterDigitLocal = 5

```

+++++
++ Field name   : TimerInterDigitRemote
++ Description  : Remote Inter digit timer, used during Remote DTMF
++             : operations
++ Valid values : 5 - 20 secs, in increment of 1 sec
++             : Default value is 5 secs
++ Note        : None
+++++

```

TimerInterDigitRemote = 5

```

+++++
++ Field name   : CallConversation
++ Description  : Used for limiting Call Conversation time.
++ Valid values : CallConversation = 0 , Calls lasts indefinite
++             : CallConversation = 1 Calls lasts for definite time
++
++ Field name   : TimerCallConversation
++ Description  : Call Conversation time, call only lasts for this time,
++             : applicable when CallConversation = 1
++ Valid values : 1 - 360 minutes in increments of 1 minute.
++             : Default value is 12 minutes
++ Note        : None
+++++

```

CallConversation = 1

TimerCallConversation = 12

```

+++++
++ Field name   : RingBackBusy
++ Description  : Enable/Disable Ringback Busy Timer.
++ Valid values : RingBackBusy = 0 , Call waits for indefinite

```

```

++ RingBackBusy = 1 Calls waits for definite time before
++ switching to next number
++
++ Field name : TimerRingBackBusy
++ Description : Ringback busy time, call switch to next number after
++ waiting for this time, applicable when RingBackBusy = 1
++ Valid values : 1 - 60 seconds in increments of 1 sec.
++ Default value is 15 seconds
++ Note : Only applicable to Outbound Round Robin Calls
+++++
```

```

RingBackBusy = 1
TimerRingBackBusy = 15

```

```

+++++
++ Field name : HangupOnSilence
++ Description : Enable/Disable Hanging of call on silence
++ Valid values : HangupOnSilence = 0 - Disable
++ HangupOnSilence = 1 - Call disconnect on silence.
++
++ Field name : TimerHangupOnSilence
++ Description : Duration after which call disconnects automatically, if no
++ voice received.
++ Valid values : 10 - 360 seconds, in increments of 1 second.
++ Default value is 30 seconds
++ Note : None
+++++
```

```

HangupOnSilence = 1
TimerHangupOnSilence = 30

```

```

+++++
++ Field name : PagingOut
++ Description : Output device for Paging messages.
++ Valid values : PagingOut = 1 - To Handsfree Speaker
++ PagingOut = 2 - To Line Level Out (Default Value)
++ Note : None
+++++
```

```

PagingOut = 2

```

```

+++++
++ Field name : PagingOutGain
++ Description : Paging output gain.
++ Valid values : 1 - 20, Default value is 20
++ Note : None
+++++
```

```

PagingOutGain = 20

```

```

+++++
++ Field name : SIPPhoneNum
++ Description : This parameter is used to define phone number for the
++ purpose of receiving incoming calls.
++ Valid values : 30 chars long, a-z,A-Z,*,0-9,.,-, No spaces allowed

```

```

++          Default is VOIP-600
++ Note     : Mandatory Parameter
++++

```

SIPPhoneNum = VOIP-600

```

++++
++ Field name  : SIPDomainName
++ Description : This parameter is used to define domain to which this
++             phone belongs to.
++ Valid values : 30 chars long, a-z,A_Z,*, 0-9,.,, No spaces allowed
++             Default is voip500.local
++ Note       : Mandatory Parameter
++++

```

SIPDomainName\_p = voip500.local  
 SIPDomainName\_s = voip500.local  
 SIPDomainName\_t = voip500.local

```

++++
++ Field name  : SIPRegister
++ Description : This parameter is used to define Phone SIP Registration.
++             If enable then Phone will try to register at given
++             registrar.
++ Valid values : SIPRegister = 0 - Disable SIP Registration (Default Value)
++             SIPRegister = 1 - Enable SIP Registration
++ Note       : For Cases when SIPRegister = 1 configure following
++             parameters.
++
++ Field name  : SIPRegistrarIP
++ Field name  : SIPRegistrarPort
++ Field name  : SIPReregistrationTime
++ Description : SIPRegistrarIP - Registrar IP address
++             SIPRegistrarPort - Registrar Port
++             SIPReregistrationTime - Registration Refresh time
++ Valid Values : SIPRegistrarPort : 1024 - 49151, 5060 (Default value)
++             SIPReregistrationTime : 10 - 14400 secs, 3600 (Def value)
++
++ Field name  : SIPRegistrarUserName
++ Field name  : SIPRegistrarUserPassword
++ Description : Optional parameters, required when using authenticating
++             Registrar.
++ Valid values : 30 chars long, a-z,A-Z,*,0-9,.,- , No spaces allowed
++ Note       : Only used when using authenticating Registrar.
++++

```

SIPRegister = 0  
 SIPRegistrarIP\_p = 0  
 SIPRegistrarIP\_s = 0  
 SIPRegistrarIP\_t = 0  
 SIPRegistrarPort= 5060  
 SIPReregistrationTime = 3600  
 SIPRegistrarUserName\_p = 0  
 SIPRegistrarUserName\_s = 0  
 SIPRegistrarUserName\_t = 0

SIPRegistrarUserPassword\_p = 0  
SIPRegistrarUserPassword\_s = 0  
SIPRegistrarUserPassword\_t = 0

```

+++++
++ Field name   : SIPOutBoundProxyIPAddress & SIPOutBoundProxyPort
++ Description  : Optional parameters. Used when using Outbound Proxy.
++ Valid values : SIPOutBoundProxyPort : 1024 - 49151, 5060 (Default value)
++
++ Field name   : SIPOutBoundProxyUserName
++ Field name   : SIPOutBoundProxyUserPassword
++ Description  : Optional parameters, required when using authenticating
++              proxy.
++ Valid values : 30 chars long, a-z,A-Z,*,0-9,.,- , No spaces allowed
++ Note        : None
+++++

```

SIPOutBoundProxyIPAddress\_p = 0  
SIPOutBoundProxyIPAddress\_s = 0  
SIPOutBoundProxyIPAddress\_t = 0  
SIPOutBoundProxyPort = 5060  
SIPOutBoundProxyUserName\_p = 0  
SIPOutBoundProxyUserName\_s = 0  
SIPOutBoundProxyUserName\_t = 0  
SIPOutBoundProxyUserPassword\_p = 0  
SIPOutBoundProxyUserPassword\_s = 0  
SIPOutBoundProxyUserPassword\_t = 0

```

+++++
++ Field name   : NumberList
++ Description  : Numbers to dial be dialed on press of Emergency and Info
++              Buttons and also on on Activation of Auxiliary Inputs.
++ Valid values : Max 30 chars, for IP Addresses, Analog Numbers, Aliases
++              and URLs
++              Example:
++              NumberList1CN1 = 10.112.71.85
++              NumberList1CN2 = 100@10.112.71.82
++              NumberList1CN3 = 1005
++              NumberList1CN4 = test@example.com
++              NumberList1CN5 =
++              NumberList1CN6 =
++ Note        : Admin can similarly configure Number List 2 to Number
++              List 7
+++++

```

NumberList1CN1 =  
NumberList1CN2 =  
NumberList1CN3 =  
NumberList1CN4 =  
NumberList1CN5 =  
NumberList1CN6 =

NumberList2CN1 =  
NumberList2CN2 =

NumberList2CN3 =  
 NumberList2CN4 =  
 NumberList2CN5 =  
 NumberList2CN6 =

NumberList3CN1 =  
 NumberList3CN2 =  
 NumberList3CN3 =  
 NumberList3CN4 =  
 NumberList3CN5 =  
 NumberList3CN6 =

NumberList4CN1 =  
 NumberList4CN2 =  
 NumberList4CN3 =  
 NumberList4CN4 =  
 NumberList4CN5 =  
 NumberList4CN6 =

NumberList5CN1 =  
 NumberList5CN2 =  
 NumberList5CN3 =  
 NumberList5CN4 =  
 NumberList5CN5 =  
 NumberList5CN6 =

NumberList6CN1 =  
 NumberList6CN2 =  
 NumberList6CN3 =  
 NumberList6CN4 =  
 NumberList6CN5 =  
 NumberList6CN6 =

NumberList7CN1 =  
 NumberList7CN2 =  
 NumberList7CN3 =  
 NumberList7CN4 =  
 NumberList7CN5 =  
 NumberList7CN6 =

```

+++++
++ Field name   : Button1CallList
++ Description  : List of numbers to be dialed from a specific List.
++ Valid values : 1 - 7. 1 (Default value)
++
++ Field name   : Button1CallNetworkPriority
++ Description  : Network priority to prioritize the Emergency Traffic.
++ Valid values : 1 - 63. 46 (Default value)
++ Note        : This field is only valid for phones having atleast 1
++              button.
+++++
  
```

Button1CallList = 1  
 Button1CallNetworkPriority = 46

```

+++++
++ Field name   : Button2Type
++ Description  : Type of Button.
++ Valid values : Button2Type = 1 - As AutoDial
++             : Button2Type = 2 - As Hookswitch (Default Value)
++
++ Field name   : Button2CallList
++ Field name   : Button2CallPriority
++ Field name   : Button2CallNetworkPriority
++ Description  : Required when Button programmed as AutoDial.
++ Description  : Required when Button programmed as AutoDial.
++ Valid values : Button2CallPriority : 2 - 10 , 2 (default value)
++             : Button2CallNetworkPriority: 1 - 63. 46 (Default value)
++ Note        : This field is only valid for phones having more than 1
++             : buttons.
+++++

```

```

Button2Type = 1
Button2CallList = 1
Button2CallPriority = 2
Button2CallNetworkPriority = 46

```

```

+++++
++ Field name   : Button3Type
++ Description  : Type of Button.
++ Valid values : Button3Type = 1 - As AutoDial
++             : Button3Type = 2 - As Hookswitch (Default Value)
++
++ Field name   : Button3CallList
++ Field name   : Button3CallPriority
++ Field name   : Button3CallNetworkPriority
++ Description  : Required when Button programmed as AutoDial.
++ Description  : Required when Button programmed as AutoDial.
++ Valid values : Button3CallPriority : 2 - 10 , 2 (default value)
++             : Button3CallNetworkPriority: 1 - 63. 46 (Default value)
++ Note        : This field is only valid for phones having more than 1
++             : buttons.
+++++

```

```

Button3Type = 5
Button3CallList = 1
Button3CallPriority = 0
Button3CallNetworkPriority = 0

```

```

+++++
++ Field name   : Button4Type
++ Description  : Type of Button.
++ Valid values : Button4Type = 1 - As AutoDial
++             : Button4Type = 2 - As Hookswitch (Default Value)
++
++ Field name   : Button4CallList
++ Field name   : Button4CallPriority
++ Field name   : Button4CallNetworkPriority
++ Description  : Required when Button programmed as AutoDial.

```



```

++ Description   : Required when Button programmed as AutoDial.
++ Valid values  : Button4CallPriority : 2 - 10 , 2 (default value)
++              : Button4CallNetworkPriority: 1 - 63. 46 (Default value)
++ Note         : This field is only valid for phones having more than 1
++              : buttons.
+++++

```

```

Button4Type = 4
Button4CallList = 1
Button4CallPriority = 0
Button4CallNetworkPriority = 0

```

```

+++++
++ Field name    : AuxIn1CallList
++ Description   : List of numbers to be dialed from a specific List.
++ Valid values  : 1 - 7, none (Disable)
++              : AuxIn1CallList = 1 (Default value)
++
++ Field name    : AuxIn1CallPriority
++ Description   : Call priority for the call originated on activation
++              : of AuxIn 1.
++ Valid values  : 2 - 10. 2 (Default Value)
++
++ Field name    : AuxIn1NetworkPriority
++ Description   : Network priority to be used for packets of the call. Used
++              : for prioritizing the traffic.
++ Valid values  : 1 - 63. 46 (Default value)
++ Note         : Admin can similarly program values for AuxIn2 and AuxIn3
+++++

```

```

AuxIn1CallList = 1
AuxIn1CallPriority = 2
AuxIn1NetworkPriority = 46

```

```

AuxIn2CallList = 1
AuxIn2CallPriority = 2
AuxIn2NetworkPriority = 46

```

```

AuxIn3CallList = 1
AuxIn3CallPriority = 2
AuxIn3NetworkPriority = 46

```

```

+++++
++ Field name    : AuxOp1ActOnHandsetCall, AuxOp2ActOnHandsetCall ,
++              : AuxOp3ActOnHandsetCall,
++              : AuxOp1ActOnBut1Call, AuxOp1ActOnBut2Call,
++              : AuxOp1ActOnBut3Call, AuxOp1ActOnBut4Call,
++              : AuxOp2ActOnBut1Call, AuxOp2ActOnBut2Call,
++              : AuxOp2ActOnBut3Call, AuxOp2ActOnBut4Call,
++              : AuxOp3ActOnBut1Call, AuxOp3ActOnBut2Call,
++              : AuxOp3ActOnBut3Call, AuxOp3ActOnBut4Call,
++
++ Description   : Auxiliary Output activation on Activation/Deactivation of
++              : Button events

```

```

++
++ Valid values : 0 - 3
++          0- Disable
++          1- Call Initiation
++          2- Call Answer
++          3- Call Termination
+++++

```

```

AuxOp1ActOnHandsetCall = 0
AuxOp1ActOnBut1Call = 1
AuxOp1ActOnBut2Call = 1
AuxOp1ActOnBut3Call = 0
AuxOp1ActOnBut4Call = 0

```

```

AuxOp2ActOnHandsetCall = 0
AuxOp2ActOnBut1Call = 1
AuxOp2ActOnBut2Call = 1
AuxOp2ActOnBut3Call = 0
AuxOp2ActOnBut4Call = 0

```

```

AuxOp3ActOnHandsetCall = 0
AuxOp3ActOnBut1Call = 1
AuxOp3ActOnBut2Call = 1
AuxOp3ActOnBut3Call = 0
AuxOp3ActOnBut4Call = 0

```

```

+++++
++ Field name : AuxOp1ActOnAuxIn1Act, AuxOp1ActOnAuxIn2Act ,
++          AuxOp1ActOnAuxIn3Act, AuxOp1ActOnAuxIn1DeAct,
++          AuxOp1ActOnAuxIn2DeAct & AuxOp1ActOnAuxIn3DeAct
++
++ Description : Auxiliary Output activation on Activation/Deactivation of
++          Auxiliary Inputs
++
++ Valid values : 0 - 1
++          0- Disable
++          1- Aux Activation when Auxiliary Input is activated or
++          deactivated
++
++          AuxOp1ActOnAuxIn1Act = 0 (Default value)
++
++ Note : Admin can similarly program values for AuxOutput 2 and 3
+++++

```

```

AuxOp1ActOnAuxIn1Act = 0
AuxOp1ActOnAuxIn2Act = 0
AuxOp1ActOnAuxIn3Act = 0

```

```

AuxOp1ActOnAuxIn1DeAct = 0
AuxOp1ActOnAuxIn2DeAct = 0
AuxOp1ActOnAuxIn3DeAct = 0

```

```

AuxOp2ActOnAuxIn1Act = 0
AuxOp2ActOnAuxIn2Act = 0

```

AuxOp2ActOnAuxIn3Act = 0

AuxOp2ActOnAuxIn1DeAct = 0  
AuxOp2ActOnAuxIn2DeAct = 0  
AuxOp2ActOnAuxIn3DeAct = 0

AuxOp3ActOnAuxIn1Act = 0  
AuxOp3ActOnAuxIn2Act = 0  
AuxOp3ActOnAuxIn3Act = 0

AuxOp3ActOnAuxIn1DeAct = 0  
AuxOp3ActOnAuxIn2DeAct = 0  
AuxOp3ActOnAuxIn3DeAct = 0

```

+++++
++ Field name   : AuxOp1ActOnRemoteKeyCode, AuxOp2ActOnRemoteKeyCode,
++             AuxOp3ActOnRemoteKeyCode
++
++ Description  : Auxiliary Output activation on specified remote keycode
++
++ Valid values : Disable,
++             Max 4 chars containing 0-9, * and #.
++             Default values are *11*, *12*, *13* correspondingly
++
++             AuxOp1ActOnRemoteKeyCode = *11* (Default value)
++             AuxOp2ActOnRemoteKeyCode = *12* (Default value)
++             AuxOp3ActOnRemoteKeyCode = *13* (Default value)
++
++ Note        : None
+++++

```

AuxOp1ActOnRemoteKeyCode = Disable  
AuxOp2ActOnRemoteKeyCode = Disable  
AuxOp3ActOnRemoteKeyCode = Disable

```

+++++
++ Field name   : AuxOp1ActOnLocalKeyCode,AuxOp2ActOnLocalKeyCode,
++             AuxOp3ActOnLocalKeyCode
++
++ Description  : Auxiliary Output activation on specified local keycode
++
++ Valid values : Disable, Max 8 chars containing 0-9, * and #.
++
++             AuxOp1ActOnLocalKeyCode = Disable (Default value)
++             AuxOp2ActOnLocalKeyCode = Disable (Default value)
++             AuxOp3ActOnLocalKeyCode = Disable (Default value)
++
++ Note        : None
+++++

```

AuxOp1ActOnLocalKeyCode = Disable  
AuxOp2ActOnLocalKeyCode = Disable  
AuxOp3ActOnLocalKeyCode = Disable

```

+++++
++ Field name   : AuxOp1DeActOnRemoteKeyCode, AuxOp2DeActOnRemoteKeyCode,
++             AuxOp3DeActOnRemoteKeyCode
++
++ Description  : Auxiliary Output deactivation on specified remote keycode
++
++ Valid values : Disable,
++             Max 4 chars containing 0-9, * and #.
++             Default values are *21*, *22*, *23* correspondingly
++
++             AuxOp1DeActOnRemoteKeyCode = *21* (Default value)
++             AuxOp2DeActOnRemoteKeyCode = *22* (Default value)
++             AuxOp3DeActOnRemoteKeyCode = *23* (Default value)
++
++ Note        : None
+++++

```

```

AuxOp1DeActOnRemoteKeyCode = Disable
AuxOp2DeActOnRemoteKeyCode = Disable
AuxOp3DeActOnRemoteKeyCode = Disable

```

```

+++++
++ Field name   : AuxOp1DeActOnLocalKeyCode,AuxOp2DeActOnLocalKeyCode,
++             AuxOp3DeActOnLocalKeyCode
++
++ Description  : Auxiliary Output deactivation on specified local keycode
++
++ Valid values : Disable, Max 8 chars containing 0-9, * and #.
++
++             AuxOp1DeActOnLocalKeyCode = Disable (Default value)
++             AuxOp2DeActOnLocalKeyCode = Disable (Default value)
++             AuxOp2DeActOnLocalKeyCode = Disable (Default value)
++
++ Note        : None
+++++

```

```

AuxOp1DeActOnLocalKeyCode = Disable
AuxOp2DeActOnLocalKeyCode = Disable
AuxOp3DeActOnLocalKeyCode = Disable

```

```

+++++
++ Field name   : AuxOp1DeActOnHandsetCall, AuxOp2DeActOnHandsetCall,
++             AuxOp3DeActOnHandsetCall,
++             AuxOp1DeActOnBut1Call, AuxOp1DeActOnBut2Call,
++             AuxOp1DeActOnBut3Call, AuxOp1DeActOnBut4Call,
++             AuxOp2DeActOnBut1Call, AuxOp2DeActOnBut2Call,
++             AuxOp2DeActOnBut3Call, AuxOp2DeActOnBut4Call,
++             AuxOp3DeActOnBut1Call, AuxOp3DeActOnBut2Call,
++             AuxOp3DeActOnBut3Call, AuxOp3DeActOnBut4Call,
++
++ Description  : Auxiliary Output 1 deactivation on call through Button #1
++             - 4 or handset.
++
++ Valid values : 0 - 3
++             0- Disable

```

```

++      1- Aux Deactivation when call is initiated through Button
++      2- Aux Deactivation when call initiated through Button is
++      connected
++      3- Aux Deactivation when call initiated through Button is
++      disconnected
++
++      AuxOp1DeActOnBut1Call = 0 (Default value)
++
++ Note   : Admin can similarly program values for AuxOutput 2 and 3

```

+++++

```

AuxOp1DeActOnHandsetCall = 0
AuxOp1DeActOnBut1Call = 0
AuxOp1DeActOnBut2Call = 0
AuxOp1DeActOnBut3Call = 0
AuxOp1DeActOnBut4Call = 0

```

```

AuxOp2DeActOnHandsetCall = 0
AuxOp2DeActOnBut1Call = 0
AuxOp2DeActOnBut2Call = 0
AuxOp2DeActOnBut3Call = 0
AuxOp2DeActOnBut4Call = 0

```

```

AuxOp3DeActOnHandsetCall = 0
AuxOp3DeActOnBut1Call = 0
AuxOp3DeActOnBut2Call = 0
AuxOp3DeActOnBut3Call = 0
AuxOp3DeActOnBut4Call = 0

```

+++++

```

++ Field name   : AuxOp1DeActOnAuxIn1Act, AuxOp1DeActOnAuxIn2Act ,
++               AuxOp1DeActOnAuxIn3Act, AuxOp1DeActOnAuxIn1DeAct,
++               AuxOp1DeActOnAuxIn2DeAct & AuxOp1DeActOnAuxIn3DeAct
++

```

```

++ Description  : Auxiliary Output deactivation on Activation/Deactivation of
++               Auxiliary Inputs
++

```

```

++ Valid values : 0 - 1
++               0- Disable
++               1- Aux Deactivation when Auxiliary Input is activated or
++               deactivated
++

```

```

++               AuxOp1DeActOnAuxIn1Act = 0 (Default value)
++

```

```

++ Note        : Admin can similarly program values for AuxOutput 2 and 3

```

+++++

```

AuxOp1DeActOnAuxIn1Act = 0
AuxOp1DeActOnAuxIn2Act = 0
AuxOp1DeActOnAuxIn3Act = 0

```

```

AuxOp1DeActOnAuxIn1DeAct = 0
AuxOp1DeActOnAuxIn2DeAct = 0
AuxOp1DeActOnAuxIn3DeAct = 0

```

AuxOp2DeActOnAuxIn1Act = 0  
 AuxOp2DeActOnAuxIn2Act = 0  
 AuxOp2DeActOnAuxIn3Act = 0

AuxOp2DeActOnAuxIn1DeAct = 0  
 AuxOp2DeActOnAuxIn2DeAct = 0  
 AuxOp2DeActOnAuxIn3DeAct = 0

AuxOp3DeActOnAuxIn1Act = 0  
 AuxOp3DeActOnAuxIn2Act = 0  
 AuxOp3DeActOnAuxIn3Act = 0

AuxOp3DeActOnAuxIn1DeAct = 0  
 AuxOp3DeActOnAuxIn2DeAct = 0  
 AuxOp3DeActOnAuxIn3DeAct = 0

```

+++++
++ Field name   : VMActionOnBut1CallInit, VMActionOnBut2CallInit ,
++             VMActionOnInboundCallsAns, VMActionOnAuxIn1Act,
++             VMActionOnAuxIn2Act & VMActionOnAuxIn3Act
++
++ Description  : Configured VM will be played to the user upon call
++               initiation by the specified button
++
++ Valid values : 1 - 5,Disable
++               Disable - Disable the field
++               1 - 5 Specifies the number of the VM to be played to the
++               user.
++
++             VMActionOnBut1CallInit = Disable (Default value)
++
++ Field name   : VMActionOnInboundCallsAns
++
++ Description  : Configuring the specified VM to start playing or stop
++               playing when the inbound calls are answered.
++
++ Valid values : 1 - 10, Disable
++               Disable - Disable the field
++               1 - 5 Specifies the number of the VM to be played to the user.
++               6 - Stops the VM #1(if playing)
++               7 - Stops the VM #2(if playing).Similarly the values 8 to 10
++               can be configured to stop the VM #3 TO VM #5
++
++ Note        : Admin can similarly program values for VMActionOnAuxIn1Act,
++               VMActionOnAuxIn2Act & VMActionOnAuxIn3Act on the guidelines
++               for the VMActionOnInboundCallsAns
+++++
  
```

VMActionOnInboundCallsAns= Disable  
 VMActionOnAuxIn1Act = Disable  
 VMActionOnAuxIn2Act = Disable  
 VMActionOnAuxIn3Act = Disable  
 VMActionOnAuxIn1DeAct = Disable

VMActionOnAuxIn2DeAct = Disable  
 VMActionOnAuxIn3DeAct = Disable  
 VMActionOnBut1CallInit = Disable  
 VMActionOnBut2CallInit = Disable  
 VMActionOnBut3CallInit = Disable  
 VMActionOnBut4CallInit = Disable

```

+++++
++ Field name   : VM1PlayOnRemoteKeyCode, VM1StopOnRemoteKeyCode
++
++ Description  : The specified VM will be played/stopped to the user upon
++                dialing the configured keycode from the remote side.
++
++ Valid values : Disable, Combination of 0-9,*,# with maximum of 4 characters
++                Disable - Disable the field
++                *12#   - can be used to either start or stop VM #1 play to
++                user when dialed from the remote side.
++
++                VM1PlayOnRemoteKeyCode = Disable(Default value)
++
++ Field name   : VM1PlayOnLocalKeyCode, VM1StopOnLocalKeyCode
++
++ Description  : The specified VM will be played/stopped to the user upon
++                dialing the configured keycode from the local keypad.
++
++ Valid values : Disable, Combination of 0-9,*,# with maximum of 8 characters
++                Disable - Disable the field
++                *123456# - can be used to either start or stop VM #1 play to
++                user when dialed from the local keypad.
++
++                VM1PlayOnLocalKeyCode = Disable(Default value)
++
++ Note        : Admin can similarly program values for VM #2 to VM #5
++                Avoid using the same keycode for start and stop of the same
++                VM.
+++++
  
```

VM1PlayOnRemoteKeyCode = Disable  
 VM1PlayOnLocalKeyCode = Disable  
 VM1StopOnRemoteKeyCode = Disable  
 VM1StopOnLocalKeyCode = Disable

VM2PlayOnRemoteKeyCode = Disable  
 VM2PlayOnLocalKeyCode = Disable  
 VM2StopOnRemoteKeyCode = Disable  
 VM2StopOnLocalKeyCode = Disable

VM3PlayOnRemoteKeyCode = Disable  
 VM3PlayOnLocalKeyCode = Disable  
 VM3StopOnRemoteKeyCode = Disable  
 VM3StopOnLocalKeyCode = Disable

VM4PlayOnRemoteKeyCode = Disable  
 VM4PlayOnLocalKeyCode = Disable  
 VM4StopOnRemoteKeyCode = Disable

VM4StopOnLocalKeyCode = Disable

VM5PlayOnRemoteKeyCode = Disable  
 VM5PlayOnLocalKeyCode = Disable  
 VM5StopOnRemoteKeyCode = Disable  
 VM5StopOnLocalKeyCode = Disable

```

+++++
++ Field name   : VM1PlayToUserInLoop
++
++ Description  : The VM #1 will be played to the user in loop
++
++ Valid values : 0 - 1
++             0- Disable, the VM play in loop
++             1- Enable, the VM play in loop
++
++             VM1PlayToUserInLoop = 0 (Default value)
++
++ Note        : Admin can similarly program values for VM #2 to VM #5
+++++
  
```

VM1PlayToUserInLoop = 0  
 VM2PlayToUserInLoop = 0  
 VM3PlayToUserInLoop = 0  
 VM4PlayToUserInLoop = 0  
 VM5PlayToUserInLoop = 0

```

+++++
++ Field name   : VMActionOnBut1CallInitRemote, VMActionOnBut2CallInitRemote
++
++ Description  : Configured VM will be played to the guard upon call
++               initiation by the specified button
++
++ Valid values : 1 - 5,Disable
++             Disable - Disable the field
++             1 - 5 Specifies the number of the VM to be played to the
++               user.
++
++             VMActionOnBut1CallInitRemote = Disable (Default value)
++
++Filed name    : VMActionOnAuxIn1ActRemote, VMActionOnAuxIn2ActRemote,
++               VMActionOnAuxIn3ActRemote
++               VMActionOnAuxIn1DeActRemote, VMActionOnAuxIn2DeActRemote,
++               VMActionOnAuxIn3DeActRemote
++Valid Value   : Disable , 1-10
++               : 1-5 (for playing of voice message #1 to #5
++               : 6-10 (for stop of voice message play #1 to #5)
++ Field name   : VMActionOnOutboundCallsAns
++
++ Description  : The specified VM will be played to the guard, once the call
++               is being answered by the the guard.
++
++ Valid values : Disable 1 - 5
++             Disable - Diable the option of playing the VM
  
```



```

++      1 - 5 Specifies the number of the VM which needs to be
++      played to the guard phone, when the call is being
++      answered by guard.
++
++      VMActionOnOutboundCallsAns = Disable (Default value)
++
++ Field name   : VM1PlayToGuardOnRemoteKeyCode, VM1StopToGuardOnRemoteKeyCode
++
++ Description  : The VM #1 will be played/stopped to the guard, once the
++                configured key is dialed by the the guard.
++
++ Valid values : Disable, Combination of 0-9,*,# with maximum of 4 characters
++                Disable - Disable the field
++                *12# - can be used to either start or stop VM #1 play to
++                       guard when dialed from the remote side.
++
++      VM1PlayToGuardOnRemoteKeyCode = Disable (Default value)
++      VM1StopToGuardOnRemoteKeyCode = Disable (Default value)
++
++ Note        : Admin can similarly program values for VM2 to VM5.
++                Avoid using the same keycode for start and stop of the same
++                VM.
++++

```

```

VMActionOnAuxIn1ActRemote = Disable
VMActionOnAuxIn2ActRemote = Disable
VMActionOnAuxIn3ActRemote = Disable
VMActionOnAuxIn1DeActRemote = Disable
VMActionOnAuxIn2DeActRemote = Disable
VMActionOnAuxIn3DeActRemote = Disable
VMActionOnHandsetOffhookRemote = Disable
VMActionOnBut1CallInitRemote = Disable
VMActionOnBut2CallInitRemote = Disable
VMActionOnBut3CallInitRemote = Disable
VMActionOnBut4CallInitRemote = Disable
VMActionOnOutboundCallsAns = Disable
VM1PlayToGuardOnRemoteKeyCode = Disable
VM1StopToGuardOnRemoteKeyCode= Disable
VM2PlayToGuardOnRemoteKeyCode = Disable
VM2StopToGuardOnRemoteKeyCode= Disable
VM3PlayToGuardOnRemoteKeyCode = Disable
VM3StopToGuardOnRemoteKeyCode= Disable
VM4PlayToGuardOnRemoteKeyCode = Disable
VM4StopToGuardOnRemoteKeyCode= Disable
VM5PlayToGuardOnRemoteKeyCode = Disable
VM5StopToGuardOnRemoteKeyCode= Disable

```

```

++++
++ Field name   : VM1PlayToGuardInLoop
++
++ Description  : The VM #1 will be played to the guard in loop
++
++ Valid values : 0 - 1
++                0- Disable, the VM play in loop
++                1- Enable, the VM play in loop

```

```

++
++          VM1PlayToGuardInLoop = 0 (Default value)
++
++ Note      : Admin can similarly program values for VM #2 to VM #5
+++++

```

```

VM1PlayToGuardInLoop = 0
VM2PlayToGuardInLoop = 0
VM3PlayToGuardInLoop = 0
VM4PlayToGuardInLoop = 0
VM5PlayToGuardInLoop = 0

```

```

+++++
++ Field name  : KeypadMode
++ Description : Mode of keypad to be used for dialing.
++ Valid values : KeypadMode = 1 - Normal dial (Default Value)
++             KeypadMode = 2 - Speed dial
++ Note        : If Keypad is in speed dial, then Admin should configure
++             the speeddial numbers (Max 30 chars) .
++             Admin can configure IP Addresses, Analog Numbers, Aliases
++             and URLs. (Max 30 chars)
+++++

```

```

KeypadMode = 1
SpeedDialKey0 =
SpeedDialKey1 =
SpeedDialKey2 =
SpeedDialKey3 =
SpeedDialKey4 =
SpeedDialKey5 =
SpeedDialKey6 =
SpeedDialKey7 =
SpeedDialKey8 =
SpeedDialKey9 =

```

```

+++++
++ Field name  : Led3
++ Description : Admin can configure Led3 Mode for specific purposes
++ Valid values : Led3 = 1 - Help on The Way
++             Led3 = 2 - Out Of Service
++
++ Note        : For Options 1 and 3, Configure following parameters
++
++ Field name  : Led3SwitchOn
++ Description : Admin can configure Led3 to switch on when remote user
++             enters specific key sequence.
++ Valid values : Max 8 chars containing 0-9, * and #.
++             Default is *98*
++ Note        : None
++
++ Field name  : Led3DurationType
++ Description : Admin can configure Led3 duration of activation.
++ Valid values : Led3Duration = 1 - For Entire Call (Default Value)
++             Led3Duration = 2 - For indefinite duration
++             Led3Duration = 3 - For Custom duration

```

```

++ Note      : If Led3Duration = 3 then configure duration time.
++
++ Field name : Led3Duration
++ Valid values : 1 - 3600 sec,in increments of 1 sec. 10 (Default Value)
++
++ Field name : Led3SwitchOff
++ Description : Admin can configure key sequence to switch off when remote
++              Admin enters specific key sequence.
++ Valid values : Max 8 chars containing 0-9, * and #.
++              Default is *97*
++ Note      : None
+++++

```

```

Led3 = 3
Led3SwitchOn = *98*
Led3DurationType = 1
Led3Duration = 10
Led3SwitchOff = *97*

```

```

+++++
++ Field name : GuardAccessCode
++ Description : Admin can set the guard access code using this field.
++ Valid values : Maximum length 8 digits(0-9).
++              GuardAccessCode = *4** (Default Guard access code)
++              It will accept the values in the following format.
++              *4*<xxxxxxx>* , x can be 0-9 or blank
++ Note      : None
+++++

```

```

GuardAccessCode = *4**

```

```

+++++
++ Field name : AuthVolAdjust , AuthSpkrMicCtrl, AuthVoiceMessage ,
++              AuthAuxOPOperation , AuthLED3Operation
++ Description : Admin can configure the authentication for different
++              operations.
++              a. Volume Adjustment During Call
++              b. Speaker and Microphone Control During Call
++              c. Voice Messages
++              d. Aux o/p Activation Deactivation
++              e. Help on the way LED activation and deactivation
++ Valid values : 0 - To disable Authentication
++              1 - To enable Authentication
++ Note      : None
+++++

```

```

AuthVolAdjust = 0
AuthSpkrMicCtrl = 1
AuthVoiceMessage = 1
AuthAuxOPOperation = 1
AuthLED3Operation = 0

```

```

+++++
++ Field name : NetworkMode
++ Description : It configures the Phone IP address Type.

```

```

++ Valid values : NetworkMode = 1 - DHCP
++           NetworkMode = 3 - STATIC (Default Value)
++ Note       : If NetworkMode = 3, must configure following
++ Field name : IPAddress
++ Field name : Netmask
++ Field name : DNSServerIP
++
++ Description : IPAddress - Phone IP Address - mandatory
++           Netmask - The Subnet Mask Address - mandatory
++           Gateway - The Default Gateway address - mandatory
++           DNSServerIP - The DNS Server IP Address - optional
++ Example:
++           NetworkMode = 3
++           IPAddress = 192.168.1.10 (Default)
++           Netmask = 255.255.255.0 (Default)
++           Gateway = 192.168.1.1 (Default)
++           DNSServerIP = (Default blank)
++
++ Note       : NetworkMode = 3 (Static) should not applied in bulk update.
++           If applied then all phones will have same Network
++           Settings.
+++++
```

```

NetworkMode = 3
IPAddress = 192.168.1.10
Netmask = 255.255.255.0
Gateway = 192.168.1.1
DNSServerIP =
```

```

+++++
++ Field name : Hostname
++ Description : It configures the unique indetifier of the phone at
++           network level
++ Valid Values : 30 chars long, a-z,A-Z,*,0-9,.,-, No space allowed.
++           Default Value is VOIP-600.
++ Note       : This Parameter should not be applied in bulk update.
++           If applied then all phones will have same hostname.
+++++
```

```

Hostname = VOIP-600
```

```

+++++
++ Field name : LogRedirection
++ Field name : LogServerIP
++ Field name : LogServerPort
++ Description : Admin can enable the logging, and route to specific
++           location.
++ Valid values : LogRedirection = 0 - To Disable Logging
++           LogRedirection = 1 - To Console
++           LogRedirection = 2 - To Network
++           LogRedirection = 3 - To File (Append)(Default Value)
++           LogRedirection = 4 - To File (New)
++ Note       : If LogRedirection = 2 must configure LogServerIP &
++           LogServerPort
++ Example:
```

```

++          LogRedirection = 2
++          LogServerIP = 10.112.71.35
++          LogServerPort = 514
++++

```

```

LogRedirection = 3
LogServerIP =
LogServerPort =

```

```

++++
++ Field name   : LogLevel
++ Description  : Admin can configure the log levels of following modules
++              of EVP
++              a. AudioLogLevel   - For Audio Module
++              b. CallControlLogLevel - For Call Control Module
++              c. CommandModLogLevel - For Webs Contact Client Module
++              d. DevCtrlLogLevel  - For Device Control Module
++              e. H323LogLevel     - For H323 Module
++              f. SIPLogLevel      - For SIP Module
++              g. TimerLogLevel    - For Timer Module
++
++ Valid values : 1 - Error
++              3 - Error + Warning (Default value)
++              7 - Error + Warning + Info
++              15 - Error + Warning + Info + Debug
++ Note        : None
++++

```

```

AudioLogLevel = 3
CallControlLogLevel = 3
CommandModLogLevel = 3
DevCtrlLogLevel = 3
H323LogLevel = 3
SIPLogLevel = 3
TimerLogLevel = 3

```

```

++++
++
++ End Of the File
++
++++

```

## Appendix F: Network Port Requirements

The network for VOIP-600 Series phones should allow the following:

- IPv4 enabled
- Allow the following protocols:
  - SIP
  - RTP
  - HTTP/HTTPS
  - SMTP
- Routed network
- DHCP Server (Optional)

### Network elements:

- SIP Proxy/Registrar
- SIP end points
- Layer-2 Switch
- WEBS Contact Server

### Ports:

The following ports need to be enabled on the VOIP-600 Series phones and allowed across the firewall and routers in the network.

Service/Module	Port	Port Type	Mode
HTTP	80	TCP/UDP	Inbound
FTP	21	TCP	Inbound
SSH	22	TCP	Inbound
Paging	5000	UDP	Inbound
SIP Signaling	5060	UDP	Inbound/Outbound
SIP Secure	5061	UDP	Inbound/Outbound
SIP Start RTP	8000 to 65535	UDP	Inbound/Outbound
STUN	3478	UDP	Inbound/Outbound
WEBS Contact Port	3001	UDP	Inbound/Outbound
SMTP Server Port	25	TCP/UDP	Inbound/Outbound