MITEL NETWORKS

5055 SIP Phone

USER GUIDE

it's about YOU

For Firmware 2.0 Revision E

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About Your SIP Phone

Welcome

Congratulations on your purchase of the Mitel Networks™ 5055 SIP Phone, an intelligent Session Initiation Protocol (SIP) telephone that manages its own call states and features. The Mitel Networks 5055 SIP Phone connects you to other SIP Phones via the Internet. You can dial by URL, IP Address, User Name, or User Number. If you have an account with a SIP Service Provider, you can also make calls to telephones on the “regular” phone network (PSTN).

The 5055 SIP Phone is a multi-line set, and can have up to three user profiles, each with its own settings.

About This User Guide

This User Guide contains information on configuring and using your 5055 SIP Phone, and is organized as follows:

- **About Your SIP Phone** (this section): basic information on the SIP Phone and its features.
- **5055 SIP Phone Features**: information on how to configure and use your SIP Phone.: information on setting up user profiles, and modifying network and SIP account configurations.
- **Appendix A — SIP Phone Interface**: overview of the SIP Phone Menu Interface that can be used to program your SIP Phone.
- **Appendix B — Web Configuration Tool**: overview of the web-based Configuration Tool that can be used to program your SIP Phone as well as to make calls.
- **Appendix C — Configuration Files:** examples of generic and specific configuration files.
- **Appendix D — Working with Firewalls**: explains how to configure the SIP phone to work with firewalls.
- **Appendix E — Working with the 3050 ICP** explains some of the benefits that can be obtained by connecting at 5055 SIP phone to Mitel Networks 3050 ICP.
- **Appendix F — Frequently Asked Questions** provides tips on how to solve some frequently encountered problems
- **Glossary**: definition of terms and acronyms found in this User Guide.

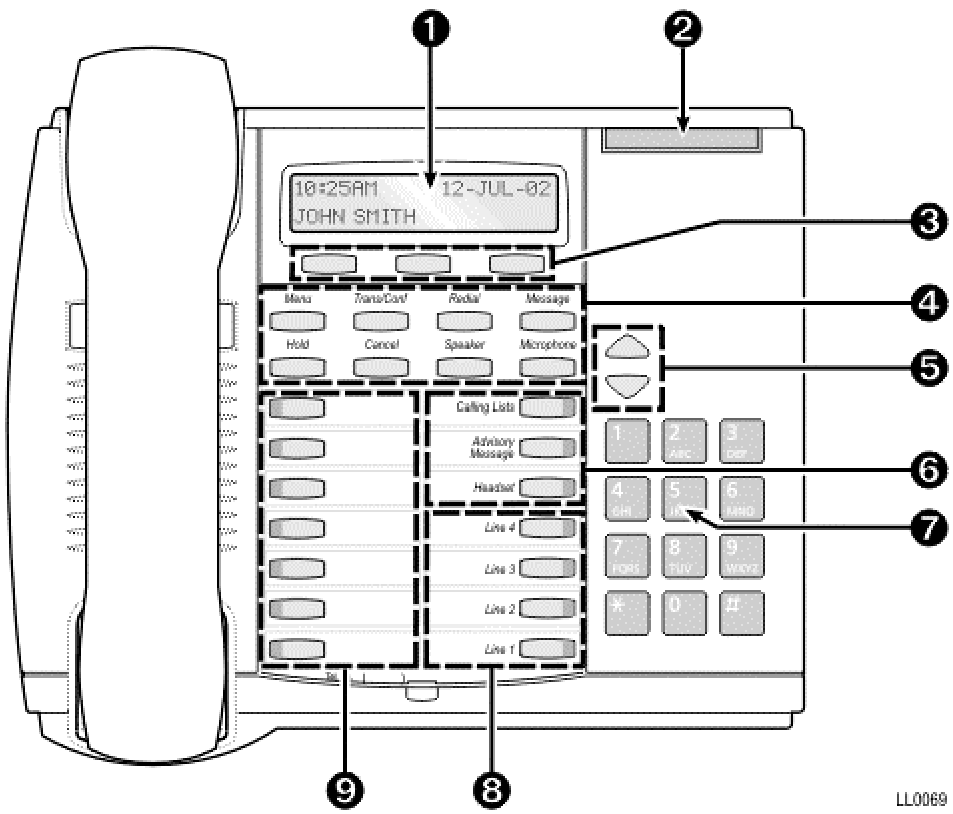
Document Conventions

- Text on the SIP Phone display or on a page of the Web Configuration Tool is shown in double quotes (for example, “CALLING LISTS?”).
- SIP Phone keys and commands on the SIP Phone display are shown in bold (for example, **Menu**).
- Sections within this document are shown in italics (for example, *Using the SIP Phone Features*).

- Links in a web page are shown as underlined text (for example, User Configuration).
- ▼ and ▲ represent the Down and Up Arrow keys on the SIP Phone (located just above the keypad).
- ► and ► represent navigating softkeys on the display.

The 5055 SIP Phone

Figure 1   5055 SIP Phone



Elements of the SIP Phone

❶ Display Screen

Provides a high-resolution, back-lit viewing area for ease of use. In default mode (default display), it shows the name of the active user (see Figure 1 above). In Menu mode, it shows prompts and information on the features.

❷ Message Waiting/Ringing Indicator Lamp

Flashes when you have an incoming call or a new message in your voice mailbox. Is on (steady) while the SIP Phone reboots.

❸ Softkeys

Select a command or choice listed on the bottom line of the display screen. These commands and choices change dynamically depending on the different modes of operation.

❹ Fixed Function Keys

Give you access to the following telephone functions:

- **Menu** (blue): Provides access to the telephone's menus.
- **Trans/Conf**: Initiates a call transfer or establishes a 3-party conference call.
- **Redial**: Redials the last number, name or address dialed.
- **Message**: Provides access to your voice mailbox (optional).
- **Hold** (red): Puts the current call on hold.
- **Cancel**: Selecting **Cancel** during a call, ends the call. When programming the SIP phone, cancels an input and returns to the previous menu level.
- **Speaker**: Initiates a handsfree call, switches between handset and handsfree mode, or disconnects a call while in handsfree mode.
- **Microphone**: Toggles the microphone off and on. In handsfree mode, a red light indicates that the microphone is ON (your party hears you). In handset and headset mode, the microphone key acts like a Mute key, and a red light indicates that the microphone is OFF (your party can't hear you).

❺ Arrow Keys

Adjust the volume of the handset, headset, or speaker, and of the ringer volume. When entering letters, changes character input from upper or lower case or vice versa. Are also used to change the display contrast, and to navigate through some menus when programming the SIP Phone. In this User Guide, the arrow keys are represented by ▼ and ▲.

❻ Fixed Feature Keys

Give you access to the following telephone features:

- **Calling Lists**: Provides immediate access to your Phone Book, Answered Calls Log, Missed Calls Log, and Outgoing Calls Log.
- **Advisory Message**: Allows you to turn your Advisory Message on or off.
- **Headset**: Allows you to enable and disable headset operation.

❼ Keypad

When making a call, used to enter the number, name, URL or IP address you want to dial. When programming the SIP Phone, used to enter information. Depending on the context, the keypad lets you enter only numbers, or numbers, letters and some special characters.

❽ Line Keys

Allow you to initiate, receive, and manage calls by using the four pre-assigned line keys. The default **Line** key is Line 1. If a line is busy, subsequent calls are received on the next available **Line** key (Line 2, Line 3, then Line 4). The **Line** keys are not assigned to a specific directory number or address (multi-line operation).

❾ Personal Keys

Provide one-touch access to programmed Speed Dial numbers.

Features of the SIP Phone

User Profiles

The 5055 SIP Phone can have up to three user profiles, including a default user profile. Each user profile has its user name and password, and can be personalized to the user's preferences. See *User Profiles* on page 10 for more information.

Administrative Mode

Some settings (network information, SIP Service Provider information, etc.) can only be modified by the system administrator, using the Administrator user name and password.

Accessing the SIP Phone's Features

You can personalize/change settings for your SIP Phone from the SIP Phone itself (SIP Phone Menu Interface) and from any personal computer connected to the Internet, using a web browser (Web Configuration Tool). You can make calls using your SIP Phone or the Web Configuration Tool. See *5055 SIP Phone* Features on page 7 for more information.

Entering Numbers and Letters Using the SIP Phone Keypad

Depending on the context, the keypad lets you enter only numbers, or numbers, letters and some special characters.

When entering letters and special characters, you rapidly press the appropriate number key several times until the desired character is displayed. Letters correspond to those on the keypad, and characters to the table below. A flashing cursor indicates the position of the character you are entering; it will advance if you press a different key on the keypad, or wait about one second.

To enter a letter in uppercase, press the ▲ key before entering the letter. Press the ▼ key to return to lowercase mode. To delete the last entered character, press the <––– softkey.

Table 1 Alphanumeric Character Entry

Dial Pad Key	Press								
	Once	Twice	3 Times	4 Times	5 Times	6 Times	7 Times	8 Times	9 Times
1	1	space	?	!	→	|			
2	2	a	b	c					
3	3	d	e	f					
4	4	g	h	i					
5	5	j	k	l					
6	6	m	n	o					
7	7	p	q	r	s				
8	8	t	u	v					
9	9	w	x	y	z				
0	0	+	&	%	$	¥	“		
*	*	.	=	:	/	;	,	–	_
#	#	@	(	)	[	]	<	>	

Accessories for the SIP Phone

The 5055 SIP Phone supports the following accessories:

Headsets

Mitel Networks has qualified a Plantronics, Inc. headset for use with the 5055 SIP Phone's dedicated headset port (Mitel part number 9132-800-500-NA). This headset is available in North America only.

The 5055 SIP Phone has a dedicated headset port (identified by the icon) at the back to connect an approved headset. An external amplifier is not needed.

Conference Units

Mitel Networks supports two conference units for use with the 5055 SIP Phone (Mitel Networks 5305 Conference Unit and Mitel Networks 5310 Conference Unit).

Figure 2 Mitel Networks Conference Unit

Important Notes

Passwords

When you first receive your 5055 SIP Phone, it has default user names and passwords for the Administrator and the default user profile. You should change these passwords as soon as possible to prevent unauthorized changes to your SIP Phone.

Table 2 Default User Names and Passwords

	Default User Name	Default Password
Administration	admin	5055
Default User Profile	user	hello

Only the Administrator can change the Administrator password. The Administrator can also change the password of all other users.

Note: The Administrator default user name cannot be changed.

Tips for Your Comfort and Safety

Don't Cradle the Handset

Prolonged use of the handset can lead to neck, shoulder, or back discomfort, especially if you cradle the handset between your ear and shoulder. If you use your 5055 SIP Phone a lot, you may find it more comfortable to use a headset.

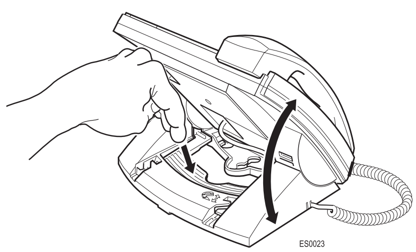
Protect your Hearing

Your 5055 SIP Phone has a control for adjusting the volume of the handset receiver or headset. Because continuous exposure to loud sounds can contribute to hearing loss, keep the volume at a moderate level.

Adjust the Phone for Easiest Viewing

- Press the tilt-release paddle on the telephone base.
- Tilt your telephone to the desired angle.
- Release the tilt-release paddle.

Figure 3 Tilt-Release Paddle



5055 SIP Phone Features

Once your SIP Phone is installed and configured (see the 5055 SIP Phone Installation Guide for more details), you can start using your phone.

This section contains information on using and personalizing your 5055 SIP Phone, and is organized as follows:

- **Accessing the SIP Phone Features**: information on how to use the Web Configuration Tool and the SIP Phone Menu Interface to access the features of your SIP Phone.
- **User Profiles**: information on user profiles, and on how to log in, log out, and activate your user profile.
- **Making and Answering Calls**: information on the basic telephony features of your SIP Phone.
- **Using the SIP Phone Features**: information on using the features of your SIP Phone.

Accessing the SIP Phone Features

You can make calls and personalize your SIP Phone from the SIP Phone itself (SIP Phone Menu Interface) or using a computer (Web Configuration Tool).

The SIP Phone Menu Interface

Most features can be directly accessed using the keys on your SIP Phone. For other features, you must use the SIP Phone Menu Interface, which is accessed using the **Menu** key (see *Appendix A — SIP* Phone Interface on page 42 for an overview of these features).

To scroll backwards or forwards through the main menu of the SIP Phone Menu Interface, press the **<<** or **>>** softkeys. To scroll forwards or backwards through the sub-menus, press the ► or ► softkeys. To go back a menu level, press the **Cancel** key. To exit the SIP Phone Menu Interface, press the **Menu** key, or go off-hook (lift handset).

In this document, procedures using the SIP Phone Menu Interface are identified by a small phone icon ().

Appendix A — SIP Phone Interface on page 42 lists all the menus and submenus available through the Sip Phone Menu Interface.

The Web Configuration Tool

You can personalize/change settings for your SIP Phone from any computer connected to the Internet using a web browser (Netscape Navigator 4 or Internet Explorer 4 (minimum), or any other equivalent browser). You can also make calls using the Web Configuration Tool.

Note: If your network is protected by a firewall, you normally will not be able to access your SIP phone via the Web Configuration Tool from outside the firewall.

In this document, procedures using the Web Configuration Tool are identified by a small computer icon ().

Appendix B — Web Configuration Tool on page 45 shows all the pages of the Web Configuration Tool, and the functions for each page.

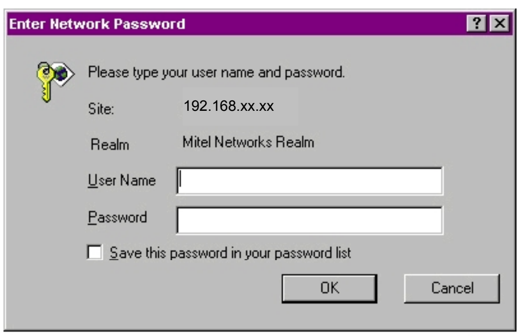
Accessing the Web Configuration Tool

To access the Web Configuration Tool:

1. Get the SIP Phone's IP address:
   - Press the **Menu** key.
   - Press the **Line 1** key on the SIP Phone. The top line of the display shows the IP address of the SIP Phone.
   - Note the IP address of the SIP Phone, and press the **Menu** key to return to the default display.

2. Launch your computer's browser.
3. Enter your SIP Phone's IP address in your browser's URL or Address field. The login screen for the Web Configuration Tool appears.

Figure 4   Web Configuration Tool Login Screen



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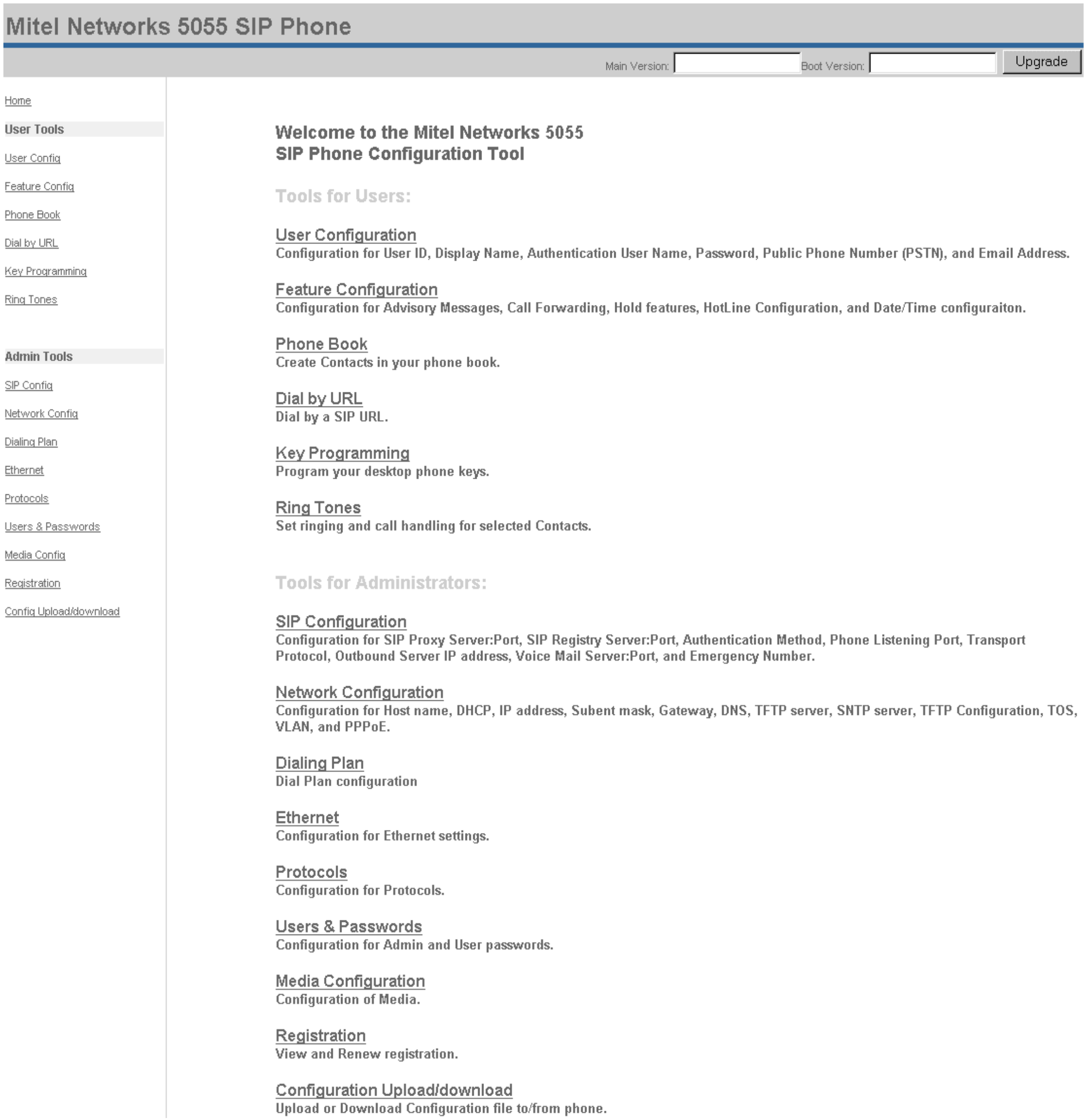
4. Enter your user profile user name and password in the appropriate fields.
5. Click the **OK** button. The home page of the Web Configuration Tool is loaded.

Figure 5   Web Configuration Tool Home Page



Refreshing Web Configuration Tool Pages

If you need to refresh a page after changing settings using the Web Configuration Tool (for example, you set your advisory message to On using the Web Configuration Tool, then turned it back to Off using the **Advisory Message** key on the SIP Phone), go to another page in the Web Configuration Tool, then come back to the page you wanted to refresh.

Do not use the **Refresh** or **Reload** button of your browser to refresh a Web Configuration Tool page after changing settings using the Web Configuration Tool. Doing so will just reapply the change you just did (and reboot your SIP phone, if you clicked a **Save and Reboot** button).

User Profiles

NOTE: By default the User Profiles are disabled for security purposes, to enable User Profiles browse to the Phones User Configuration screen, locate the MultiUser Profile option and select On. Select Save & Reboot to enable MultiUser Profiles.

To access all the features the 5055 SIP Phone has to offer, you need a user profile and must be registered with a SIP Service Provider (you can use the SIP Phone without a user profile, but will not be able to use all its features). A user profile is usually created and registered by the system administrator.

Once you have a user profile, you can personalize the following:

- your user profile information (password, display name),
- your user profile feature settings (**Personal** keys, phone book, call answer settings, etc.).

Your SIP Phone supports up to three user profiles, including a default user profile. The default user profile is always logged in, and cannot be deleted.

When you **log in** the SIP Phone, you are automatically registered with your SIP Service Provider, and can receive calls on the SIP Phone. When you **activate** your user profile, the SIP Phone uses your user profile preferences (Speed Dial keys, etc.), and you can use the Web Configuration Tool to make calls or change your user profile settings. More than one user can be logged in at the same time, but only one user profile can be active at a time.

Note: Only users who have a user profile defined on a SIP Phone can log in to that SIP Phone.

You can also temporarily register with your SIP Service Provider on a SIP Phone that does not have your user profile. While you are temporarily registered on that phone, you can make and receive calls with the SIP Phone, but cannot use the Web Configuration Tool.

Logging In and Out

Like a personal computer, the 5055 SIP Phone allows different users to log in and access their personal settings. Incoming calls addressed to the logged-in user's name, SIP URL or Number will be delivered to that SIP Phone. So, for example, if you need to do some work in the lab but still want to answer incoming calls that you would normally answer at your desk, you can log in to the lab's 5055 SIP Phone.

Note: Logging in on your SIP Phone automatically registers you with your SIP Service Provider (assuming you have a SIP Service Provider, and a user profile defined on the SIP Phone). If you have a SIP Service Provider, and for any reason the registration process fails, “*NO REG*” appears on your display.

Logging In

1. Press the **Menu** key.
2. Press the **>>** softkey. “USERS?” is displayed.
3. Press the **OK** softkey. “1.LOGIN?” is displayed.
4. Press the **OK** softkey.
5. Enter your user profile user name, and press the **Submit** softkey.

6. Enter your user profile password, and press the **Submit** softkey (if your SIP Service Provider does not require a password, enter any character, and delete it using the <––– softkey before pressing the **Submit** softkey).
   - If you made a mistake while entering your user name or password, “LOGIN UNSUCCESSFUL” is displayed. Press the **Retry** softkey to return to step 4, or the **Cancel** softkey to return to step 3.
7. “ACTIVATE PROFILE?” is displayed. To log in and activate your user profile, press the **Yes** softkey. To log in without activating your user profile, press the **No** softkey.
8. “SET LOGIN EXPIRY?” is displayed.
   - If you don’t want a login expiry time, press the **No** softkey, and continue with step 10.
   - If you want to set your user profile to automatically be logged out (de-registered) after a given period, press the **Yes** softkey and continue with the next step.
9. “LOGIN EXPIRY (HR)” is displayed.
   - To enter a value in hours, enter the value using the keypad, and press the **Submit** softkey.
   - To enter a value in minutes, press the **Minute** softkey, enter the value using the keypad, and press the **Submit** softkey.
   - To enter a value in days, press the **Days** softkey, enter the value using the keypad, and press the **Submit** softkey.
10. Once you are logged in, “LOGIN SUCCESSFUL” is displayed. Press the **OK** softkey, then the **Menu** key to return to the default display.

Logging Out

If you have an account with a SIP Service Provider, logging out of the SIP Phone automatically de-registers you with your SIP Service Provider.

1. Press the **Menu** key.
2. Press the **>>** softkey. “USERS?” is displayed.
3. Press the **OK** softkey. “1.LOGIN?” is displayed.
4. Press the ► softkey. “2.LOGOUT?” is displayed. Press the **OK** softkey.
5. Enter your user profile user name, and press the **Submit** softkey.
6. Enter you user profile password, and press the **Submit** softkey (if your SIP Service Provider does not require a password, enter any character, and delete it using the <––– softkey before pressing the **Submit** softkey).
7. Your user name is displayed. Press the **LogOut** softkey.
8. “LOGOUT CONFIRMED” is displayed. Press the **OK** softkey, then the **Menu** key to return to the default display.

Activating a Profile

When you activate your user profile, the SIP Phone uses your preferences (Speed Dial keys, Display Name, etc.), and you can access the Web Configuration Tool to make calls or change your user profile settings.

Note: To activate your user profile, you must already be logged in the SIP Phone.

1. Press the **Menu** key.
2. Press the **>>** softkey. “USERS?” is displayed.
3. Press the **OK** softkey.
4. Press the ▶ softkey until “3.ACTIVATE PROFILE?” is displayed. Press the **OK** softkey.
5. The name of the active user profile is displayed. Press the **Change** softkey to activate a different user profile.
6. Enter your user profile user name, and press the **Submit** softkey.
7. Enter you user profile password, and press the **Submit** softkey (if you’re SIP Service Provider does not require a password, enter any character, and delete it using the <––– softkey before pressing the **Submit** softkey).
8. Your user name is displayed. Press the **Yes** softkey to activate your profile.
9. “PROFILE ACTIVATED” is displayed. Press the **OK** softkey, then the **Menu** key to return to the default display.

Temporary Registration

Temporary registration tells your SIP Service Provider that you can temporarily receive calls at that phone.

To temporarily register on a SIP Phone, you need the following information from your SIP Service Provider:

- Registration user name and password
- SIP Service Provider server IP address
- Phone IP address
- Registration method

Registering on the SIP Phone

1. Get the IP address of the SIP Phone to which you are registering:
   - Press the **Menu** key.
   - Press the **Line 1** key on the SIP Phone. The top line of the display shows the IP address of the SIP Phone.
   - Note the IP address of the SIP Phone, and press the **Menu** key to return to the default display.
2. Press the **Menu** key.
3. Press the **>>** softkey. “USERS?” is displayed.
4. Press the **OK** softkey.
5. Press the ▶ softkey until “6.REGISTRATION?” is displayed. Press the **OK** softkey.

6. For **ENTER USERID:** enter your SIP Registration user name, and press the **Submit** softkey.
7. For **ENTER PASSWORD:** enter your SIP Registration password, and press the **Submit** softkey (if your SIP Service Provider does not require a password, enter any character, and delete it using the <—— softkey before pressing the **Submit** softkey).
8. For **CONTACT IP ADDREESS:** enter the SIP Phone's IP address, and press the **Submit** softkey.
9. For **SERVER ADDRESS**: enter your SIP Service Provider server address, and press the **Submit** softkey.
10. For **TO ADDRESS:** enter <user>@<SIP server address>, and press the **Submit** softkey.

11. For **REGISTRATION METHOD**: select the registration method by pressing the appropriate softkey:
    - **None**: no registration authentication.
    - **Basic**: **Authentication**
    - **Digest**: **Authentication**
12. The display shows the registration method chosen. Press the **Submit** softkey to confirm your choice, or the **Cancel** softkey to choose another registration method (step 11).
13. Enter the registration duration (in hours), and press the **Submit** softkey (after that duration, you will automatically be de-registered).
14. The display shows "REGISTER NOW". Press the **Confirm** softkey to register, or the **Cancel** softkey to change your registration duration (step 13).

    When registration is complete, the display shows "REGISTRATION SUCCESSFUL".

Making and Answering Calls

This section shows you how to make and receive calls with the SIP Phone. Only basic call making procedures are shown (that is, calls made using the keypad). For information on other call making features see *Using the SIP Phone Features* on page 17.

Your SIP Phone can be used in any of the three following modes:

- **Handset mode**: this is when you are using the handset to talk and listen to your party.
- **Headset mode**: this is when you have a headset connected to your SIP Phone, and you use it to talk and listen to your party. Headset mode is activated by pressing the **Headset** key (you can have a headset connected to your SIP Phone and still use the handset to make your calls).
- **Handsfree mode**: this is when you are using the SIP Phone's handsfree speaker to talk and listen to your party. Handsfree mode is activated by pressing the **Speaker** key.

This section shows how to make and receive call in any of the three modes, as well as how to change from one mode to another. In the rest of the document, instructions are given for the handset mode only, for clarity's sake.

Making Calls

With the 5055 SIP Phone you can dial:

- by number (user number, telephone number).
- by name (user name).
- by URL (SIP URL address, SIP IP address).

Note: To dial a regular telephone number, you must have a SIP Service Provider that provides access to the regular (PSTN) phone network.

When the connection is successful, the address of your party is displayed (truncated to 14 characters), and a counter starts at the top right of the display. If the line is busy, the display shows “BUSY HERE”.

To end or abort a call:

- All modes:
  - Press the **Cancel** key or the **Hangup** softkey to get a new dial tone.
  - Press the **Line** key associated with the call to return to the default display.
- Handset mode only:
  - Put the handset back in its cradle.
- Handsfree mode only:
  - Press the **Speaker** key. This returns you to the default display.

Dialing by Number

To dial by number:

1. Get a dial tone:
   - Handset mode: lift the handset.
   - Headset mode, press a **Line** key. You can also start entering a number after pressing the **Headset** key (you won’t get a dial tone).
   - Handsfree mode: press the **Speaker** key *or* press a **Line** key. You can also start entering a number after pressing the **Speaker** key (you won’t get a dial tone).

   This selects the first free line (Line 1 if all lines are free). To select another line, press the associated **Line** key. The light of the selected **Line** key turns red. In handsfree mode, the **Microphone** key light turns red; in headset mode, and the **Headset** key light turns red.
2. Enter the number of the party you want to reach using the keypad. 5055 SIP. Phones that are connected to 3050 ICPs, can use abbreviated two or three-digit numbers to reach other 3050 ICP attached phones, or regular 7-digit numbers to reach phones on the PSTN. Check with your phone system installer to learn if your system supports these options.
   - If you mistype a number, press the <—— softkey to delete it, and re-enter the correct number.
   - To delete all characters entered and enter a different number, press the **Cancel** softkey.
3. Press the **Dial** softkey. The number you entered is dialed, and the light of the selected **Line** key turns green.

Dialing by Name

To dial by name:

1. Get a dial tone:
   - Handset mode: lift the handset.
   - Headset mode,: press a **Line** key.
   - Handsfree mode: press the **Speaker** key *or* press a **Line** key.

   This selects the first free line (Line 1 if all lines are free). To select another line, press the associated **Line** key. The light of the selected **Line** key turns red. In handsfree mode, the **Microphone** key light turns red; in headset mode, and the **Headset** key light turns red.
2. Press the **Name** softkey.
3. Using the keypad, enter the name of the party you want to reach (see *Entering Numbers and Letters Using the SIP Phone Keypad* on page 4 for information on entering letters and symbols).
   - If the name has more than 20 characters, the display will only show the rightmost 20 characters.
   - If you mistype a character, press the <—— softkey to delete it, and re-enter the correct character.
   - To delete all characters entered and enter a different name, press the **Cancel** softkey.
4. Press the **Dial** softkey. The name you have entered is dialed, and the light of the selected **Line** key turns green.

Dialing by SIP URL

To dial a URL:

1. Get a dial tone:
   - Handset mode: lift the handset.
   - Headset mode press a **Line** key.
   - Handsfree mode: press the **Speaker** key *or* press a **Line** key.

   This selects the first free line (Line 1 if all lines are free). To select another line, press the associated **Line** key. The light of the selected **Line** key turns red. In handsfree mode, the **Microphone** key light turns red; in headset mode, and the **Headset** key light turns red.
2. Press the **URL** softkey.
3. Using the keypad, enter the address of the party you want to reach (see *Entering Numbers and Letters Using the SIP Phone Keypad* on page 4 for information on entering letters and symbols).
   - If the URL has more than 20 characters, the display will only show the rightmost 20 characters.
   - If you mistype a character, press the <—— softkey to delete it, and re-enter the correct character.
   - To delete all characters entered and enter a different address, press the **Cancel** softkey.
4. Press the **Dial** softkey. The address you have entered is dialed, and the light of the selected **Line** key turns green.

Answering Calls

An incoming call will ring on the first available line (Line 1 if all lines are free). If all lines are bus, and Call Forward on Busy is not enabled the caller gets a busy signal (refer to Enabling/Disabling Call Forward for details on call forwarding). While the phone is ringing, the Ringing indicator flashes red, the name of the caller is displayed and the associated **Line** key light flashes green.

To answer an incoming call:

- Handset mode:
  - Lift the handset.
- Headset mode:
  - Press the **Headset** key and press the flashing green **Line** key.
- Handsfree mode:
  - Press the **Speaker** key, or
  - Press the associated **Line** key (this will activate the handsfree mode).

When you answer the call, a counter starts at the top right of the display.

Switching between Handset, Headset and Handsfree Modes

With the 5055 SIP Phone, you can make and receive calls using the attached handset, the handsfree speaker, or an approved headset (see *Headsets* on page 5 for information on approved headsets).

Switching Between Handset and Handsfree Modes

- To go from handset to handsfree mode:
  - Press the **Speaker** key.
  - Put the handset back in its cradle.
  - When you are connected, the **Microphone** key light turns red. You can now talk to your party using the handsfree speaker.
- To go from handsfree to handset mode:
  - Lift the handset. The **Microphone** key light turns off.
  - You can now talk to your party using the handset.

Switching Between Handset and Headset Modes

- To go from handset to headset mode:
  - Press the **Headset** key. The **Headset** key light turns red.
  - Put the handset back in its cradle.
  - You can now talk to your party using the headset.
- To go from headset to handset mode:
  - Lift the handset.
  - Press the **Headset** key. The **Headset** key light turns off.
  - You can now talk to your party using the handset.

Switching Between Headset and Handsfree Modes

- To go from headset to handsfree mode:
  - Press the Headset Key, followed by the **Speaker** key. The **Headset** key light turns off.
  - When you are connected, the **Microphone** key light turns red. You can now talk to your party using the handsfree speaker.
- To go from handsfree to headset mode:
  - Press the **Headset** key. The **Headset** key light turns red, and the **Microphone** key light turns off.
  - You can now talk to your party using the headset.

Using the SIP Phone Features

The features in this section can be accessed using the Web Configuration Tool and/or the SIP Phone Menu Interface. You cannot change these settings using the SIP Phone Menu Interface while on a call. You can change these settings using the Web Configuration Tool while on a call, but the changes will not take effect until you've finished your current calls.

Note: Some Web Configuration Tool settings require you to reboot your SIP Phone. If you click the **Save and Reboot** button while on a call, you will lose the connection when your phone reboots.

Advisory Messages

Setting up an Advisory Message

You can set up an advisory message to alert callers to your current status (for example, when you're going on vacation). You set up advisory messages using the Web Configuration Tool.

Note: You cannot change your Advisory Message settings while on a call. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).

1. Click Feature Configuration.
2. Turn Advisory message **On i**n the pull-down menu (you can use the Advisory message key on the phone to toggle this on or off too).
3. Choose the message you want to display from the drop-down menu at the right of "Advisory Message:".
   - If none of the choices suit you, select the message "Other reason", and enter the desired message beside "Other:" (any message longer than 20 characters will be truncated on the SIP Phone display).
4. Click the **Apply** button at the bottom of the web page. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

Enabling/Disabling your Advisory Message

When your Advisory Message is on, the Advisory Message indicator key turns red, and your Advisory Message periodically replaces the time and date on your SIP Phone display.

Note: You cannot change your Advisory Message settings while on a call.

Using the SIP Phone Interface

- Press the **Advisory Message** key to activate/deactivate your Advisory Message.

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. To enable the advisory message, select **On** in the drop-down menu at the far right of "Advisory Message:". To disable the advisory message, select **Off**.
4. Click the **Apply** button at the bottom of the web page. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

Call Forward

Call Forward lets you redirect incoming calls to an alternate number:

- Call Forward Always redirects all incoming calls regardless of the state of your telephone.
- Call Forward No Answer redirects calls after the programmed number of rings if you don't answer.
- Call Forward Busy redirects calls when all four lines are busy.

The default setting is Call Forward Off for all three options. You can set two or more Call Forward options On at the same time.

Note: When Call Forward is active, "*FWD ON*" alternates with the date on the SIP Phone's display.

Note: You cannot change these settings while on a call.

Setting Up Call Forward

You can set your Call Forward settings using the Web Configuration Tool or the SIP Phone Menu Interface.

Note: You cannot change your Call Forward settings while on a call.

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. You can enable Call forwarding by either:
   - setting the Call Forwarding field to **On**, and entering the Forwarding Address in the associated field. This can be the URL of another SIP phone, URL of a SIP voicemail account or a PSTN number (provided that the SIP server supports PSTN gateway functions).

   or

   - setting the Call Forwarding field to **On** , and leaving the Forwarding Address blank. In this case, the phone will forward the call to voicemail automatically. This will work provided you have the URL of your SIP voicemail account programmed into the Voice Mail Server field of the SIP Configuration Page.

4. For Call Forward No Answer, enter how many times the phone will ring before the call is forwarded.
5. Click the **Apply** button. A confirmation screen is displayed.
6. Click the **OK** button. Your SIP Phone is updated.

Using the SIP Phone Menu Interface

1. Press the **Menu** key.
2. Press the **>>** softkey until "FEATURE CONFIG?" is displayed.
3. Press the **OK** softkey. "CALL FORWARDING?" is displayed.
4. Press the **OK** softkey. "FWD ALWAYS:" is displayed, with its status beside it ("*ON*" or "*OFF*").
    - If you do not need to change your Call Forward Always settings, go to step 10.
    - To program your Call Forward Always settings, continue below.
5. Press the **Review** softkey. The display shows the current forwarding address.
    - If the address is blank and a valid voicemail server URL has been programmed into SIP Configuration, the SIP Phone will forward calls to your voice mailbox
6. Press the **Program** softkey to change the address to which the call will be forwarded.
7. To forward your calls to your voice mailbox, press the **Yes** softkey and continue with step 10. To forward your calls to another address, press the **No** softkey and continue with the next step.
8. Enter the address where your calls will be forwarded.
    - To enter a name, press the **Name** softkey before entering any characters.
    - To enter a URL, press the **URL** softkey before entering any characters.
9. Press the **Submit** softkey.
10. Press the **Next** softkey. "FWD NO ANSWER:" is displayed, with its status beside it ("*ON*" or "*OFF*")
    - If you do not need to change your Call Forward No Answer settings, go to step 14.
11. Press the **Options** softkey to change the number of rings before a call is forwarded.
12. Enter the number of rings (from 0 to 9) with the keypad, and press the **Save** softkey.
13. To program your other Call Forward No Answer settings, repeat steps 5 to 9, then continue below.
14. Press the **Next** softkey. "FWD BUSY:" is displayed, with its status beside it ("*ON*" or "*OFF*")
    - To program your Call Forward Busy settings, repeat steps 5 to 9, then continue below.
    - If you do not need to change your Call Forward Busy settings, continue below.
15. Press the **Exit** softkey, then the **Menu** key to return to the default display.

Enabling/Disabling Call Forward

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click <u>Feature Configuration</u>.

3. For each type of Call Forward (Always, No Answer, Busy), change its status (On or Off) using the drop down menu beside its name.
4. Click the **Apply** button at the bottom of the web page. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

Using the SIP Phone Menu Interface

1. Press the **Menu** key.
2. Press the **>>** softkey until “FEATURE CONFIG?” appears, and press the **OK** softkey.
3. “1.CALL FORWARDING?” is displayed. Press the **OK** softkey.
4. The display shows the status of Call Forward Always at the top right (“*ON* or *OFF*). Press the **TurnOn** softkey to activate call Forward Always, or the **TurnOff** softkey to deactivate it.
5. Press the **Next** softkey.
6. The display shows the status of Call Forward No Answer at the top right (“*ON* or *OFF*). Press the **TurnOn** softkey to activate Call Forward No Answer, or the **TurnOff** softkey to deactivate it.
7. Press the **Next** softkey.
8. The display shows the status of Call Forward Busy at the top right (“*ON* or *OFF*). Press the **TurnOn** softkey to activate Call Forward Busy, or the **TurnOff** softkey to deactivate it.
9. Press the **Exit** softkey, then the **Menu** key to return to the default display.

Call Transfer

You can transfer an active call to another party. To do so, at least one line must be free on the SIP Phone.

Note: The 5055 SIP phone supports 4 lines. If all lines are busy on your phone, none of your callers will be able to transfer their call away from you to another phone. You must first free up one of the lines to allow callers to transfer a call away from your phone.

Transferring a Call to an Unconnected Third Party

1. While on a call, press the **Trans/Conf** key. The call is put on hold.
2. Press a free **Line** key, a Speed dial key or redial.
3. Call the party to whom you want to transfer the call.
    - If you want to talk to this person, wait until the connection is established then press the **Trans** softkey to transfer the held call (attended call transfer).
    - If you don’t need to talk to this person, press the **Trans** softkey immediately, and then hang up. The held call will be transferred to the call in progress, even if it has not yet been picked up (blind or unattended call transfer).
    - If you want to cancel the transfer, press the **Cancel** softkey. You are returned to the held call.

Transferring a Call to a Third Party Already on Hold

1. While on a call, press the **Trans/Conf** key. The call is put on hold.
2. Press the **Line** key of the call on hold to which you want to transfer the call, and press the **Trans** softkey. You can then hang up.

Call Waiting

You can have up to four active calls on your SIP Phone. Any new call goes to the next free line; if all lines are busy, the caller gets a busy signal.

When a new call comes in, you hear a call waiting tone, the name of the new caller is displayed, and the next available **Line** key light flashes green.

- To answer the incoming call while already connected to another call, press the **Line** key of the incoming call. The current call will be put on hold, and you are connected to the new caller (see *Putting a Call on Hold* on page 24 for information on dealing with calls on Hold).

Calling List Logs

The Calling Lists Logs keeps a record of your answered, missed and outgoing calls. It records the five most recent calls for each of the three types of calls. For example, the five most recent incoming calls are logged while the five most recent missed calls are logged. The most recent call appears at the top of the each log.

The call information recorded includes the party's number, name or URL address, the call duration, and the time and date of each call.

When you have missed incoming calls, the number of calls missed replaces the date on the SIP Phone display.

Note: The calling lists log information is stored directly in the SIP Phone. Your user profile must be logged in and active to use your Calling List Logs.

Viewing the Calling List Logs

To view information on an incoming, missed, or outgoing Calling List entry:

1. Press the **Calling Lists** key (you can also get to the Calling List by pressing the **Menu** key, then the **>>** softkey until you reach "CALLING LISTS?", then the **OK** softkey).
2. Scroll using the ▶ and ▶ softkeys to the desired log (Missed Calls, Answered Calls or Outgoing Calls), and press the **OK** softkey.
3. The display shows how many calls are in that log. Use the ▼ and ▲ keys to view the calls in the log.
4. For each call, you can:
   - View the information about that call (press the **Detail** softkey, then the **Done** softkey to return).
   - Delete the call from the Calling List (press the **Delete** softkey and follow the prompts).
   - Dial the caller's address (press the **Dial** softkey. This exits the call log, and the SIP Phone returns to the default display at the end of the call).

5. To view calls from another log, press the **Cancel** key, then the **OK** softkey, and repeat steps 2 to 4. When you are finished, press the **Menu** key to return to the default display.

Conference Call (3-Way)

The 5055 SIP Phone supports three-party conferences.

Note: Conference call is not available when your SIP Phone is set for G.729 audio codec (the **Conf** softkey is replaced by **NA** to indicate that the feature is not available).

Adding a Third Party to a Call in Progress

1. Press the **Trans/Conf** key. The call is put on hold.
2. Press a free **Line** key.
3. Enter the address of the new party and press the **Dial** softkey.
4. Once you have connected with the new party, press the **Conf** softkey. The call on hold is connected to the call in progress.

   Note: If the new party does not answer, press the **Cancel** key twice to return to the held party.

Adding a Party on Hold to a Call in Progress

1. Press the **Trans/Conf** key. The call is put on hold.
2. Press the **Line** key of the party already on hold.
3. Once you have connected with the new party, press the **Conf** softkey. The call put on hold in step 1 is connected to the call in progress.

Leaving a Conference Call

To leave a Conference Call, hang up the handset, press the **Hangup** softkey, or press the **Cancel** key.

Note: If the originator of the conference call hangs up, then the other two parties do not remain connected. If either of the called parties hangs up, the call will remain connected.

Display Contrast

You can adjust the contrast of the display to suit your preference.

Changing the Display Contrast

- While the phone is idle, use the ▼ or ▲ key to adjust the display contrast to the desired level (press repeatedly to change by more than one level).

Note: When changing the display contrast this way, the setting is first stored in temporary memory. The temporary memory is saved to permanent (flash) memory at regular intervals during the day. If your SIP Phone loses power or reboots between the time you changed the setting and a flash memory update, the new setting will be lost.

Changing the Display Contrast—Immediate Save

1. Press the **Menu** key.

2. Press the **>>** softkey until “PHONE SETTINGS?” appears, and press the **OK** softkey.
3. Press the ► softkey until “3.LCD CONTRAST?” appears, and press the **OK** softkey.
4. Use the ▼ or ▲ key to adjust the display contrast to the desired level (press repeatedly to change by more than one level), and press the **OK** softkey. The new setting is saved in the permanent (flash) memory.
5. You are returned to the Phone Settings menu. To return to the default display, press the **Menu** key.

Display Name

The Display Name is the name displayed on your SIP Phone when you are logged in and your user profile is active. That name also appears on the display of a call’s recipient. You can change your display name using the Web Configuration Tool.

Note: You cannot change your Display Name while on a call.

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click User Configuration.
3. Beside “User Display name”, enter the name you want to appear on the display.
4. Click the **Save and Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated, and then reboots.

Do Not Disturb

Do Not Disturb forwards all your calls directly to your voice mailbox, so you are not disturbed by a ringing phone. If you do not have a voice mailbox setup, the callers will get a busy signal.

Note: When Do Not Disturb is active, “*DND ON*” alternates with the date on the SIP Phone’s display (if both Call Forward and Do Not Disturb are on, “*DND ON*” alternates with the time on the display).

Note: You cannot change your Do Not Disturb settings while on a call.

Activating/Deactivating Do Not Disturb

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. Select **On** or **Off** from the drop down menu beside “Do Not Disturb”.
4. Click the **Apply** button. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

Using the Phone Menu Interface

1. Press the **Menu** key.
2. Press the **>>** softkey until “FEATURE CONFIG?” is displayed, and press the **OK** softkey.
3. Press the ► softkey. “2.DO NOT DISTURB?” is displayed.

4. Press the **OK** softkey. The current status of Do Not Disturb is displayed at the top right (“*ON*” or “*OFF*”).
5. Press the **TurnOn** softkey to activate Do Not Disturb, or the **TurnOff** softkey to deactivate it.
6. Press the **Exit** softkey, then the **Menu** key to return to the default display.

Hold

You can have up to four calls on hold at the same time on your SIP Phone.

Putting a Call on Hold

- To place a call on Hold, press the **Hold** key; the associated **Line** key flashes red while its call is on hold. To retrieve a call from Hold, press the associated **Line** key.

Changing On Hold Settings

When you place a call on hold, you will get a regular beep after a programmed delay to remind you that you have a call on hold (if the handset is in its cradle, you will hear the beep through the handsfree speaker). When another party puts you on hold, you hear a regular beep to remind you that you are on hold; you can turn off this beep if desired.

Note: You cannot change your Hold settings while on a call.

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. To remove the regular beep you hear when you are on hold, select **Off** in the drop-down menu beside “Beep on Hold”.
4. To define the delay before your SIP Phone reminds you that you have a caller on hold, enter a value in seconds beside “Held call will ring back after:”.
5. Click the **Apply** button. A confirmation screen is displayed.
6. Click the **OK** button. Your SIP Phone is updated.

Muting a Call

To mute your SIP Phone so the person on the other end of the line cannot hear you, press the **Microphone** key. To turn off the Mute function, press the **Microphone** key once more.

- In handset and headset modes, the **Microphone** key light is red while the call is muted.
- In handsfree mode, the **Microphone** key light is off while the call is muted.

Password

You can change your user profile password using the SIP Phone Menu Interface, or the Web Configuration Tool.

Note: If you have an account with a SIP Service Provider, use the password given to you by the SIP Service Provider.

Note: You cannot change your password while on a call.

Changing Your Password

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click User Configuration.
3. Change the password beside “Password:”.
4. Click the **Save and Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated, and then reboots.

Using the SIP Phone Menu Interface

1. Press the **Menu** key.
2. Press the **>>** softkey. “USERS?” is displayed.
3. Press the **OK** softkey.
4. Press the ► softkey until “4.CHANGE PASSWORD?” is displayed, and press the **OK** softkey.
5. Enter your user profile user name, and press the **Submit** softkey.
6. Enter your current password, and press the **Submit** softkey (if your existing password is blank, enter any character, and delete it using the <—— softkey before pressing the **Submit** softkey).
7. Enter your new password, and press the **Submit** softkey.
8. Enter your new password again, and press the **Submit** softkey.
   - If you have entered both instances of the new password correctly, “NEW PASSWORD CREATED” is displayed. Press the **OK** softkey.
   - If you have made a mistake, “PASSWORD MISMATCH” is displayed. Press the **Retry** softkey to go back to step 7.
9. Press the **Menu** key to return to the default menu.

Personal Keys

Using the SIP Phone Menu Interface, you can program a **Personal** key as a Speed Dial key.

You change the feature of a **Personal** key by deleting the existing programming and applying a new program (see the entry for the actual feature for instructions on programming and using **Personal** Keys with that feature).

Note: You cannot change your **Personal** Keys settings while on a call.

Verifying a Personal Key’s Program

1. Press the **Menu** key.
2. Press the **>>** softkey until “PROGRAM MEMORY KEYS?” appears, and press the **OK** softkey.
3. Press the **Personal** key you want to check. The key’s light turns red.
   - If the key is not yet programmed, the display reads “UNUSED KEY”.
   - If the key is already programmed, its associated program is displayed.

4. Press the **Menu** key to return to the default display.

Deleting a Personal Key's Program

1. Press the **Menu** key.
2. Press the **>>** softkey until "PROGRAM MEMORY KEYS?" appears, and press the **OK** softkey.
3. Press the **Personal** key you want to clear. The key's light turns red, and its associated programming is displayed.
4. Press the **Delete** softkey.
5. "DELETE ITEM?" is displayed. Press the **YES** softkey to delete it.
6. "UNUSED KEY" is displayed. To return to the default menu, press the **Menu** key.

Phone Book

Your user profile's Phone Book can hold up to five contacts.

Note: You cannot change your Phone Book settings while on a call.

Creating/Modifying a Phone Book

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Dial by Phone Book.
3. For each contact, enter a nickname, and the SIP address for this contact (name, number or URL).
   - To change a contact, simply type over an existing one.

   Note: When entering a telephone number, enter it without any separators.
4. Click the **Apply** button to save your contacts. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

Making Calls With the Phone Book

Using the SIP Phone:

To make a call to a contact on your Phone Book:

1. Get a dial tone (see *Making Calls* on page 14).
2. Press the **Calling Lists** key.
3. "1.PHONE BOOK?" is displayed, Press the **OK** softkey.
4. Use the ▼ and ▲ keys to go to the contact you want to call, and press the **Dial** softkey. The contact's address is dialed.

Using the Web Configuration Tool:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Dial by Phone Book.
3. Select the contact you want to reach in the drop-down menu beside "Select Contact:".

4. Click **Dial**. The address is dialed (in handsfree mode) on the next available **Line** key (Line 1 if all lines are free).
   - If you want to use the handset or headset, lift the handset or press the **Headset** key before clicking “Dial” in the Web Configuration Tool.
   - If you want to use another line than Line 1, press the desired line key on the SIP Phone before clicking “Dial” in the Web Configuration Tool.

Redial

Redial calls back the last party you dialed using the SIP Phone or the Web Configuration Tool.

Note: Your SIP Phone will not remember the last call dialed if it loses power.

1. Get a dial tone (see *Making Calls* on page 14).
2. Press the **Redial** key. The last number/name/address you called (or tried calling) is dialed.

   Note: Pressing the **Redial** key without lifting the handset will automatically put you in handsfree (speaker) mode.

Ringer Pitch and Volume

You can change the ringer volume using the SIP Phone Menu Interface, or by pressing the ▼ or ▲ key on the SIP Phone while the phone is ringing (one key press per level). You can also change the pitch of the ringer by using the SIP Phone Menu Interface.

Note: The ringer settings are specific to the SIP Phone, not to user profiles. You cannot change your ringer settings while on a call.

1. Press the **Menu** key.
2. Press the **>>** softkey until “PHONE SETTINGS?” appears, and press the **OK** softkey.
3. Press ► until “2.RINGER SOUNDS?” appears, and press the **OK** softkey.
4. “SET RINGER VOLUME?” is displayed.
   - If you don’t want to change the ringer volume, press the **No** softkey and go to step 7.
   - If you want to change the ringer volume, press the **Yes** softkey and continue below.
5. The phone starts ringing. Use the ▼ or ▲ key to adjust the volume to the desired level (one key press per level), and press the **Submit** softkey.
6. “SET RINGER VOLUME?” is displayed. Press the **No** softkey.
7. “SET RINGER PITCH?” is displayed.
   - If you don’t want to change the ringer pitch, press the **No** softkey to return to the Phone Settings menu, or the **Menu** key to return to the default display.
   - If you want to change the ringer pitch, press the **Yes** softkey and continue below.
8. The phone starts ringing. Use the ▼ or ▲ key to adjust the pitch to the desired level (one key press per level), and press the **Submit** softkey.
9. “SET RINGER PITCH?” is displayed. Press the **No** softkey to return to the Phone Settings menu, or the **Menu** key to return to the default display.

Speaker Volume (Handset, Headset and Handsfree)

To change the volume of the handset, headset or handsfree speaker volume:

Note: The time and date are specific to the SIP Phone, not to user profiles.

1. Get a dial tone (see *Making Calls* on page 14).
2. Press the ▼ key to decrease the volume, or the ▲ key to increase the volume (one key press per level).
3. Put the SIP Phone on-hook (replace handset on cradle, press the **Headset** key, or press the **Speaker** key).

The new setting will stay in effect until you change it again (if the SIP Phone loses power, the settings will return to the factory default settings).

Speed Dial

You can program a **Personal** Key with Speed Dial, so you call someone with one key press.

Programming a Speed Dial Key

1. Press the **Menu** key.
2. Press the **>>** softkey until “PROGRAM MEMORY KEYS?” appears, and press the **OK** softkey.
3. Press the **Personal** key you want to program. The key’s light turns red.
   - If the key is not yet programmed, the display reads “UNUSED KEY”.
   - If the key is already programmed, its associated feature is displayed. You must delete a key’s programming before you can apply a new one (press the **Delete** softkey and follow the prompts).
4. Press the **AddNew** softkey.
5. “ENTER NUMBER” is displayed.
   - If you want to enter a name address, press the **Name** softkey.
   - If you want to enter a URL address, press the **URL** softkey.
6. Enter the number (or name, or URL address), and press the **Save** softkey.
7. “KEY SAVED” is displayed. Press the **OK** softkey.
8. “PROGRAM MEMORY KEYS?” is displayed. Press the **OK** softkey to program more function keys, or the **Menu** key to return to the default display.
9. To add a label beside a **Personal** key you programmed:
   - Lift the plastic protector using the tab at the bottom.
   - Write the information on the card below the plastic protector.
   - Put the card and plastic protector back on the phone (insert top first).

Editing a Speed Dial Key

1. Press the **Menu** key.
2. Press the **>>** softkey until “PROGRAM MEMORY KEYS?” appears, and press the **OK** softkey.
3. Press the **Personal** key you want to edit. The key’s light turns red, and its associated feature is displayed.
4. Press the **Edit** softkey.
5. The current number, name or URL is displayed. Press the <––– softkey to delete the characters, starting from the rightmost character, and type in the new number, name or address.
6. Press the **Save** softkey.
7. “KEY SAVED” is displayed. Press the **OK** softkey.
8. “PROGRAM MEMORY KEYS?” is displayed. Press the **OK** softkey to program more function keys, or the **Menu** key to return to the default display.

Making Calls Using Speed Dial

To make a call using a personal key programmed with Speed Dial:

1. Get a dial tone (see *Making Calls* on page 14).
2. Press the **Personal** key programmed with the desired Speed Dial number/name/address. The key’s number/name/address is dialed.

Time and Date

You can change the date and time using the SIP Phone Menu Interface, or the Web Configuration Tool. Usually, your SIP Phone gets its time and date from an SNTP server (see *Modifying the Network Configuration* in the section), and all you need to do is adjust your time zone twice a year if your area uses Daylight Savings Time. If you don’t have an SNTP server, you will need to set the time and date manually.

Note: The time and date are specific to the SIP Phone, not to user profiles. You cannot change your time and date settings while on a call.

Adjusting your Time Zone

SNTP servers usually provide Greenwich Mean Time data. To adjust the time and date for your area, you need to specify your time zone (if your area uses daylight Savings Time, you will need to adjust this twice a year):

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Network Configuration.
3. In the “Additional Servers” section, beside “Time Zone:”, enter the difference between your time zone and the GMT, adjusting for Daylight Savings Time as needed (see *Time Zones* on page 58 for a table of world time zones versus GMT).
4. Click the **Save an Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated, and then reboots.

Changing the Time and Date

Use this procedure only if no SNTP server is provided. You will need to reprogram these settings every time the phone reboots.

With the Web Configuration Tool:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. Enter your date and time. Use the day-month-year format for the date and the 24-hour clock format for the time.
4. Click the **Apply** button. A confirmation screen is displayed.
5. Click the **OK** button. Your SIP Phone is updated.

With the SIP Phone Menu Interface:

1. Press the **Menu** key.
2. Press the **>>** softkey until “PHONE SETTINGS?” appears.
3. Press the **OK** softkey. “1.TIME/DATE” is displayed.
4. Press the **OK** softkey. “SET TIME?” is displayed, with the currently programmed time.
   - If you don’t need to change the time, press the **No** softkey and go to step 9 to change the date.
   - If you need to change the time, press the **Yes** softkey and continue below.
5. “12 or 24 HR FORMAT?” is displayed. Press the **12** softkey if you want to enter the time in am/pm, or the **24** softkey to enter the time using the 24-hour clock format.
6. Enter the time (for example, 1236 for 12:36, or 220 for 2:20), and press the **Submit** softkey.
7. If you are entering the time using am/pm, press the **AM** or the **PM** softkey.
8. “SET TIME?” is displayed, with the new time. Press the **No** softkey to set the date.
9. “SET DATE?” is displayed.
   - If you don’t need to change the date, press the **No** softkey to return to the Phone Settings menu, or the **Menu** key to return to the default display.
   - If you need to change the date, press the **Yes** softkey and continue below.
10. Enter the date (for example, enter 161202 for 16 December 2002), and press the **Submit** softkey.
11. “SET DATE?” is displayed. Press the **No** softkey to return to the Phone Settings menu, or the **Menu** key to return to the default display.

Web Dialing

You can make calls using the Web Configuration Tool. You can dial by URL, or using the Phone Book (see *Making Calls With the Phone Book* on page 26 for information on the latter).

To dial by URL:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Dial by URL.
3. Enter the URL of the party you want to reach.
4. To make a call:
    - Handset mode: lift the handset, and click the **Dial** button on the Web Configuration Tool.
    - Headset mode: press the **Headset** key, and click the **Dial** button on the Web Configuration Tool.
    - Handsfree mode: click the **Dial** button on the Web Configuration Tool.

    The call is made on the first available line (Line 1 if all lines are free). To use a specific **Line** key, press the desired **Line** key before clicking the **Dial** button on the web page.

Administrator Tools

This section contains the following information on configuring the administrative settings of the 5055 SIP Phone:

- Changing Passwords
- Setting Up User Profiles
- Creating/Modifying a SIP Account
- Modifying the Network Configuration
- Upgrading the Firmware of the SIP Phone
- Additional Settings

These settings are changed using the Web Configuration Tool or the SIP Phone Menu Interface.

Note: You cannot change these settings while on a call.

Changing Passwords

The Administrator can change the password for all defined users using the Web Configuration Tool or the SIP Phone Menu Interface.

Note: The Administrator password should be changed as soon as possible to prevent unauthorized access to the Administrator functions of the SIP Phone.

Using the Web Configuration Tool

1. Access the Web Configuration Tool using the Administrator user name and password (see *The Web Configuration Tool* on page 7).
2. Click Security Config.
3. Change the passwords as required (note: you cannot change user names using this screen).
4. Click the **Apply** button. A confirmation screen is displayed.
5. Click the **OK** button. The SIP Phone is updated.

Using the SIP Phone Menu Interface

You must repeat this procedure for each password that needs to be changed.

1. Press the **Menu** key.
2. Press the **>>** softkey. “USERS?” is displayed.
3. Press the **OK** softkey.
4. Press the ► softkey until “4.CHANGE PASSWORD?” is displayed, and press the **OK** softkey.
5. Enter the Administrator user name, or the user name for the user profile whose password you are changing, and press the **Submit** softkey.

6. Enter the current password, and press the **Submit** softkey (if the existing password is blank, enter any character, and delete it using the <––– softkey before pressing the **Submit** softkey).
7. Enter the new password, and press the **Submit** softkey.
8. Enter the new password again, and press the **Submit** softkey.
    - If you have entered both instances of the new password correctly, “NEW PASSWORD CREATED” is displayed. Press the **OK** softkey.
    - If you have made a mistake, “PASSWORD MISMATCH” is displayed. Press **Retry** to go back to step 7.
9. Press the **Menu** key to return to the default menu.

Setting Up User Profiles

Your SIP Phone can have two personalized user profiles, in addition to the default profile. Each profile stores information about the associated user as well as personalized configurations. Users access their profile with a user name and password.

Viewing a Profile’s User Name

To view the active user profile’s user name:

1. Press the **Menu** key.
2. Press the **Line 3** key. The display shows the user profile display name (top) and user name (bottom).
3. Press the **Menu** key to return to the default display.

Creating a User Profile

1. Press the **Menu** key.
2. Press the **>>** softkey. “USER?” is displayed.
3. Press the **OK** softkey.
4. Press the ► softkey until “5.MANAGE PROFILES?” is displayed.
5. Press the **OK** softkey.
6. Enter the Administrator user name, and press the **Submit** softkey.
7. Enter the Administrator password, and press the **Submit** softkey.
8. Press the ▼ key until your reach a vacant user profile (profile 1 is the default user profile).
9. Press the **AddNew** softkey.
10. Enter a user name for this new user profile, and press the **Submit** softkey.
11. Enter a password for this new user profile, and press the **Submit** softkey (if the password is blank, enter any character, and delete it using the <––– softkey before pressing the **Submit** softkey).
12. Enter the password again, and press the **Submit** softkey.

13. Enter the name that will appear on the phone display when the new user profile is active, and press the **Submit** softkey.
14. Enter the user's SIP Server Authentication name, and press the **Submit** softkey.
15. "NEW PROFILE CREATED" is displayed. Press the **OK** softkey.
16. To create another user profile, press **Exit**, and repeat this procedure from Step 4. To exit the procedure and return to the default display, press the **Menu** key.

Modifying a User Profile

The following user profile information can be added/modified using the Web Configuration Tool:

- User ID/Extension
- SIP Authentication User Name
- SIP Authentication Password
- Public (PSTN) Phone Number
- E-Mail Address

Note: The user profile to be modified must be logged in and active.

To enter/change that information:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click User Configuration.
3. Enter/change the information as needed.
4. Click the **Save and Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. The SIP Phone is updated and reboots.

Deleting a User Profile

1. Press the **Menu** key.
2. Press the **>>** softkey. "USER?" is displayed.
3. Press the **OK** softkey.
4. Press the ► softkey until "5.MANAGE PROFILES?" is displayed.
5. Press the **OK** softkey.
6. Enter the Administrator user name, and press the **Submit** softkey.
7. Enter the Administrator password, and press the **Submit** softkey.
8. Press the ▼ key until your reach the user profile you want to delete (profile 1 is the default user profile, and cannot be deleted).
9. Press the **Remove** softkey.
10. "REMOVE USER PROFILE?" is displayed. Press the **Confirm** softkey to delete the user profile.
11. Press **AddNew** to create a new user profile (see *Creating a User Profile* above), the ▼ or ▲ key to go to another user profile, or the **Menu** key to return to the default display.

Creating/Modifying a SIP Account

You can modify the SIP account information of the SIP Phone using the Web Configuration Tool.

1. Access the Web Configuration Tool.
2. Click SIP Configuration.
3. Enter/change the SIP Account information as needed (see Table 11 on page 54 for more information on these settings).
4. Click the **Save and Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. The SIP Phone is updated and reboots.
6. For each user profile, enter/change the associated SIP Authentication user name and password (see *Modifying a User Profile* on page 34 for more information).

Modifying the Network Configuration

Viewing the IP and MAC Addresses

To view the Internet Protocol (IP) address and the Media Access Control (MAC) address of the SIP Phone:

1. Press the **Menu** key.
2. Press the **Line 1** key. The IP and MAC addresses are displayed.
3. Press the **Menu** key to return to the default display.

Modifying Network Configurations

You can modify the following network configuration settings of the SIP Phone using the Web Configuration Tool or the SIP Phone Menu Interface (see Table 12 on page 57 for more information on these settings).

- Basic Settings:
  - SIP Phone host and domain names (web tool only).
  - DHCP status.
  - Address type.
  - SIP Phone IP address and subnet mask (supplied automatically by ISP/LAN if DHCP is on).
  - Default gateway (supplied automatically by ISP/LAN if DHCP is on).
  - Primary and secondary DNS addresses (supplied automatically by ISP/LAN if DHCP is on).
- Additional Servers Settings:
  - TFTP server.
  - SNTP server and time zone of SIP Phone.

- Advanced Settings:
  - TFTP configuration (allows configuration of SIP Phones using configuration files; see *Configuration Files* on page 38 for more information).
  - Type of Service and 802.1 Priority (Quality of Service parameters).
  - Virtual LAN ID.
  - PPPoE status and PPPoE login name and password.

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Network Configuration.
3. Add/update the information as needed.
4. Click the **Save and Reboot** button. A confirmation screen is displayed.
5. Click the **OK** button. The SIP Phone is updated and reboots.

Using the SIP Phone Menu Interface

1. Press the **Menu** key.
2. Press the **>>** softkey until “PHONE SETTINGS?” is displayed, and press the **OK** softkey.
3. Press the ► softkey until “4.NETWORK SETTINGS?” is displayed.
4. Press the **OK** softkey.
5. “DHCP” is displayed with its current status (“*ON*” or “*OFF*”).
6. If needed, press the **TurnOff** softkey to disable DHCP, or the **TurnOn** softkey to enable it.
7. Press the **Next** softkey until the next parameter you want to change is displayed.
8. Press the **Review** softkey to view its current setting.
   - If you need to change the value, press the **Change** softkey, enter the new value, then press the **Submit** softkey (when the entry can only be an IP address, pressing * twice, rapidly - enters a period).
   - To leave the value as it is, press the **Exit** softkey.
9. Repeat steps 7 and 8 until all the desired changes have been made.
10. Press the **Exit** softkey, then the **Menu** key to return to the default display.
11. For the settings to take effect, you must restart your SIP Phone:
    - When you are back in the default display, press the **Menu** key.
    - Press *, then **0** on the keypad. The SIP Phone restarts.

Upgrading the Firmware of the SIP Phone

The phone uses TFTP to download firmware upgrades from a TFTP server. There are two methods that can be used to do this: one uses the SIP phone's softkey menu system to perform the upgrade, the other uses the upgrade button on the main phone configuration web-page. The methods function differently:

- With web-page upgrade, the phone's original configuration is preserved, so that it will function as it did prior to upgrade, without the need to reconfigure settings.
- With soft-key menu upgrade, the phone's previous configuration may not be saved and restored – which would necessitate manual re-load of a previously saved configuration file or manual re-configuration of phone parameters (refer to Configuration Upload/Download Page).

Viewing the Firmware Version

Using the Web Configuration Tool

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7). If the Web Configuration Tool is currently opened, click Home.
2. The main and boot firmware versions are displayed near the top of the home page.

Using the SIP Phone Menu Interface

1. Press the **Menu** key.
2. Press the **Line 2** key. The main and boot firmware versions are displayed.
3. Press the **Menu** key to return to the default display.

Upgrading the firmware using the web-page Interface

You upgrade the SIP Phone by downloading the new firmware from the appropriate TFTP server (the TFTP server is programmed with the Network Configuration).

CAUTION: **During this procedure,** DO NOT **remove power from the SIP Phone while firmware is downloading or the phone is rebooting. This may result in severe damage to your SIP Phone.**

1. Log into the phone configuration web page.
2. Click the **Upgrade** button.
3. The Firmware Upgrade page is displayed. If the TFTP server URL is OK as is, click either the **Upgrade** or the **Upgrade & Set Factory Defaults** button. If you chose to reset to factory defaults, you may have to reset the phone's settings manually, or if a recent copy of the phone configuration file exists, you may be able to restore all but the user passwords in one easy, step. (refer to Configuration Upload/Download Page).
4. If you selected **Upgrade & Set Factory Defaults** in step 3, you will have to re-enter the SIP Authorization password, and possibly the PPPoE and User Profile passwords using the softkey menus or the web page, as these parameter are lost.

Upgrading the firmware using the SIP Phone Menu Interface

You upgrade the SIP Phone by downloading the new firmware from the appropriate TFTP server (the TFTP server is programmed with the Network Configuration).

CAUTION: **During this procedure,** DO NOT **remove power from the SIP Phone while firmware is downloading or the phone is rebooting. This may result in severe damage to your SIP Phone.**

1. To restart the phone, press the **Menu**, ∗ and 0 keys in sequence.
2. When the display shows “Booting…”, press and hold the **2** key on the keypad. This upgrades the boot firmware of the SIP Phone.
3. When “UPGRADE FIRMWARE?” appears, release the key and press the **YES** softkey.
4. The phone now gives you the option of using the displayed TFTP IP address (this IP address is set in the Network Configuration Page), or entering one of your own.
5. Press the ▼ key. The firmware starts downloading.
6. When the new firmware has finished downloading, the SIP Phone reboots. This process may take a minute or two. It’s complete when the display shows a time and date on the top line of the display (default display). In some cases, upgrade of the boot firmware will automatically trigger upgrade of the Main firmware. If you need to force the upgrade of the Main firmware, press the **Menu**, ∗ and 0 keys in sequence, then hold the **1** key (the phone then requires you to select the TFTP IP address, and press the ▼ key). This process may take a minute or two, and is complete when the display shows a time and date on the top line of the display (default display).
7. In some cases it may be necessary to reset the settings of the SIP Phone to factory defaults (note: this procedure will erase all your settings then replace them by the factory defaults). To do this
   - Press the **Menu** key.
   - Press ***** on the keypad.
   - Press **#** on the keypad.
   - Press and hold the **3** key on the keypad until “USE FACTORY DEFAULTS?” is displayed.
   - Press the **YES** softkey. The settings are reset, and the phone reboots.

   When the time and date are displayed, you must reprogram the SIP Phone or reload a saved configuration file (see *Changing Passwords*, *Setting Up User Profiles*, *Creating/Modifying a SIP Account* and Modifying the Network Configuration in this section). Saved configuration files do not restore passwords, so you will have to re-enter your passwords manually.

Additional Settings

Configuration Files

The 5055 SIP Phone supports configuration files for automatic programming of the phones. There are two types of configuration files:

- **Generic**: a generic configuration file applies the settings defined in it to all the SIP Phones.

- **Specific**: a specific configuration file applies the settings defined in it to a specific SIP Phone.

Configuration files are stored on the TFTP server, and are downloaded by the SIP Phones connecting to that server every time the phones reboot. The generic configuration file is loaded first, then the specific configuration file. If both files contain settings for the same parameter, the specific configuration file will overwrite the information from the generic configuration file.

Before a SIP Phone can automatically download configuration files from a TFTP server, it must have the following Network Configuration settings configured via the web configuration tool:

- TFTP Server address (in Additional Servers section)
- TFTP Configuration = Yes (in Advanced section)

When a SIP Phone with these settings reboots, it starts by requesting the generic configuration file from the TFTP Server. If the file exists, it is downloaded and all of the parameters in it overwrite the existing settings for these parameters on the SIP Phone. Then, the SIP Phone requests its specific configuration file from the TFTP Server. If the file exists, it is downloaded and all of the parameters in it overwrite the existing settings for these parameters on the SIP Phone (including those from the specific configuration file, if applicable). Only the parameters defined in the configuration files are overwritten on the SIP Phone. If the SIP Phone requests a configuration file that is not on the TFTP server, no settings are changed on the phone.

Note: When a SIP Phone uses configuration files, you can still change settings manually, but if these settings are also defined in the configuration files, the configuration file will overwrite the manual settings the next time the SIP Phone reboots.

Generic Configuration File (SIPGeneric.cfg)

Used to change global settings such as Media Configuration, Voice Mail server, etc. The generic configuration file is a text file saved as “SIPGeneric.cfg” on the TFTP server. You can create a generic configuration file by using a text application such as Notepad or SimpleText, or by using your favorite word processing application and saving the file as a text file. *Example of a Generic Configuration File* in Appendix C shows all the possible settings you can have in a generic configuration file.

Specific Configuration File (SXXXXXXXXXXXX.cfg)

Used to change phone-specific settings such as user profiles, Hot Line configuration, etc. Each specific configuration file (one per SIP Phone) is a text file saved as “SXXXXXXXXXXXX.cfg” on the TFTP server, where the Xs are the 12-character hexadecimal MAC address of the SIP Phone. You can create a generic configuration file by using a text application such as Notepad or SimpleText, or by using your favorite word processing application and saving the file as a text file.

Example of a Specific Configuration File in Appendix C shows all the possible settings you can have in a specific configuration file.

Hot Line

When a hot line number/address is set up, the SIP phone automatically dials that number/address when it goes off-hook (handset lifted, etc.).

The Hot Line number is programmed using the Web Configuration Tool:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Feature Configuration.
3. To activate the Hot Line, select **On** in the drop down menu beside “Hot Line Mode”.
4. You can enter a number or URL address for the Hot Line.
    - To enter a number, select **NUM_MODE** in the drop down menu beside “Address Type:”.
    - To enter a URL, select **URL_MODE** in the drop down menu beside “Address Type:”.
5. Enter the number/URL of the Hot Line beside “Destination Address:”.
6. Click the **Apply** button. A confirmation screen is displayed.
7. Click the **OK** button. The SIP phone is updated.

The programming steps listed above let the caller over-ride the pre-programmed hot line number by dialing their own. If you need to program the phone to block all other outgoing calls, other than those to the Hot line number, you need to add this rule to the Dialing Plan:

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
xx		2			

This rule recognizes, and blocks all outgoing manual dial attempts, permitting only hot line dialing. You could create a plan to allow “911” calls, but block all other outgoing calls:

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
911	0	0			
xxx	0	3			

Media Configuration

You can change the following Media Configuration settings using the Web Configuration Tool:

- Audio codec type and frame size.
- DTMF type and payload type.

To change media configuration settings:

1. Access the Web Configuration Tool (see *The Web Configuration Tool* on page 7).
2. Click Media Configuration.
3. Change the information as needed. See Table 19 on page 67 for more information on these settings.

Note: If you set the audio codec type to G.729, the users will not be able to use the Conference Call feature.

4. Click the **Apply** button. A confirmation screen is displayed.
5. Click the **OK** button. The SIP Phone is updated.

Resetting to Factory Defaults

If needed, you can erase all the settings of a SIP Phone using the SIP Phone Interface:

1. Press the **Menu** key.
2. Press the ***** key on the keypad.
3. Press the **#** key on the keypad.
4. Press and hold the **3** key on the keypad until "USE FACTORY DEFAULTS?" is displayed.
5. Press the **YES** softkey. The settings are reset, and the phone reboots.

Appendix A — SIP Phone Interface

The SIP Phone Menu Interface is used to view or program a number of the SIP Phone and user profile settings.

SIP Phone Menu Interface

The SIP Phone Menu Interface is accessed by pressing the **Menu** key. To navigate between items of the main menu, press the **>>** and **<<** softkeys. To navigate between the items of the sub-menus, use the ▶ and ▶ softkeys.

Table 3 SIP Phone Menu Interface Settings

Main Menu	Sub-Menu	Notes
USERS?	1.LOGIN?	To log in a user profile.
	2.LOGOUT?	To log out a user profile.
	3.ACTIVATE PROFILE?	To activate a user profile.
	4.CHANGE PASSWORD?	To change a password.
	5.MANAGE PROFILES?	To add/delete a user profile.
	6.REGISTRATION? *Enter UserID* *Enter Password* *Contact IP Address* *Server Address* *To Address* *Registration Method* *Reg'n Expiry (HR)*	Temporarily registers with a SIP Service Provider. *SIP Authorization. User Name* *SIP Authorization. Password* *Typically: user_name@server* *SIP registration server IP Address* *None, Basic or Digest*
CALLING LISTS?	1.PHONE BOOK?	To make calls using phone book entries (as programmed with the Web Configuration Tool).
	2.MISSED CALLS?	Log of missed calls. You can make calls from this log.
	3.ANSWERED CALLS?	Log of answered calls. You can make calls from this log.
	4.OUTGOING CALLS?	Log of outgoing calls made from your 5055 SIP Phone. You can make calls from this log.
FEATURE CONFIG?	1.CALL FORWARDING? *Always, No Answer, Busy*	To program/enable/disable call forward.
	2. DO NOT DISTURB?	To enable/disable Do Not Disturb feature.
PROGRAM MEMORY KEYS?	*Speed Dial*	To program **Personal** keys.
PHONE SETTINGS?	1. TIME/DATE?	To change the time and date.
	2. RINGER SOUNDS? *Volume, Pitch*	To change the ringer pitch and volume.
	3. LCD CONTRAST	To change the LCD contrast (immediate save).

Table 3 SIP Phone Menu Interface Settings (continued)

Main Menu	Sub-Menu	Notes
PHONE SETTINGS? (con't)	4. DEVICE PARAMETERS? *Software Version* *MAC Address*	To view the software version and MAC address of the SIP Phone.
	5. PROTCOL CONFIG? *HTTP* *TFTP* *TELNET*	To enable or disable protocol settings.
	6. MULTI USER CONFIG?	Turns the MultiUser profiles On or Off. User profiles work best with stand-alone installations (not directly connected to a 3050 ICP). Turn this feature on, only if your SIP phone is stand-alone, and you need this feature.
	7. LANGUAGE? *en_CA* *fr_CA* *fr_FR* *en_US* *en_GB* *en_AU* *es_MX* *es_US*	Changes the phone display language English Canadian French Canadian French France English USA English Great Britain English Australian Spanish Mexican Spanish USA
	8. RING TONES? *CA* *US* *GB* *DE* *NL* *AU* *NZ* *MX* *FR*	Changes the phone tone plan. Canada USA Great Britain Germany Netherlands Australia New Zealand Mexico France
	9. NETWORK SETTINGS? *DHCP* *Phone IP Address* *Phone Subnet Mask* *Default Gateway* *Outbound Server, Port* *SIP Proxy Server, Port* *SIP Proxy Port Num* *Voice Mail Server, Port* *Primary/Secondary DNS Servers* *TFTP, SNTP Servers* *Eth.Autoneg*	To change Network settings (see Table 12 on page 57 for more information on these settings). Enables/disables Ethernet auto-negotiation

Menu Key Commands

There are a few other settings you can access using the SIP Phone's **Menu** key.

Menu + Line Key Commands

1. These commands give you access to information about your SIP Phone. To access them, press the **Menu** key, then press the appropriate **Line** key. Press the **Menu** key to return to the default display.

Table 4 Menu + Line Key Commands

Display	Menu + Line 1	Menu + Line 2	Menu + Line 3
Top Line	SIP Phone IP address	Main software version	Display name
Bottom Line	SIP Phone MAC address	Boot software version	User name

Menu + Keypad Commands

These commands perform actions on the SIP Phone. To access them, press the Menu key, then each of the keypad keys in succession.

- To restart the SIP Phone: **Menu**, *****, **0**.
- To reset the SIP Phone to factory defaults: **Menu**, *****, **#**, hold **3**.

Caution: Resetting the SIP Phone to factory defaults will erase all the programming on the SIP Phone, and replace it by the factory default settings.

Appendix B — Web Configuration Tool

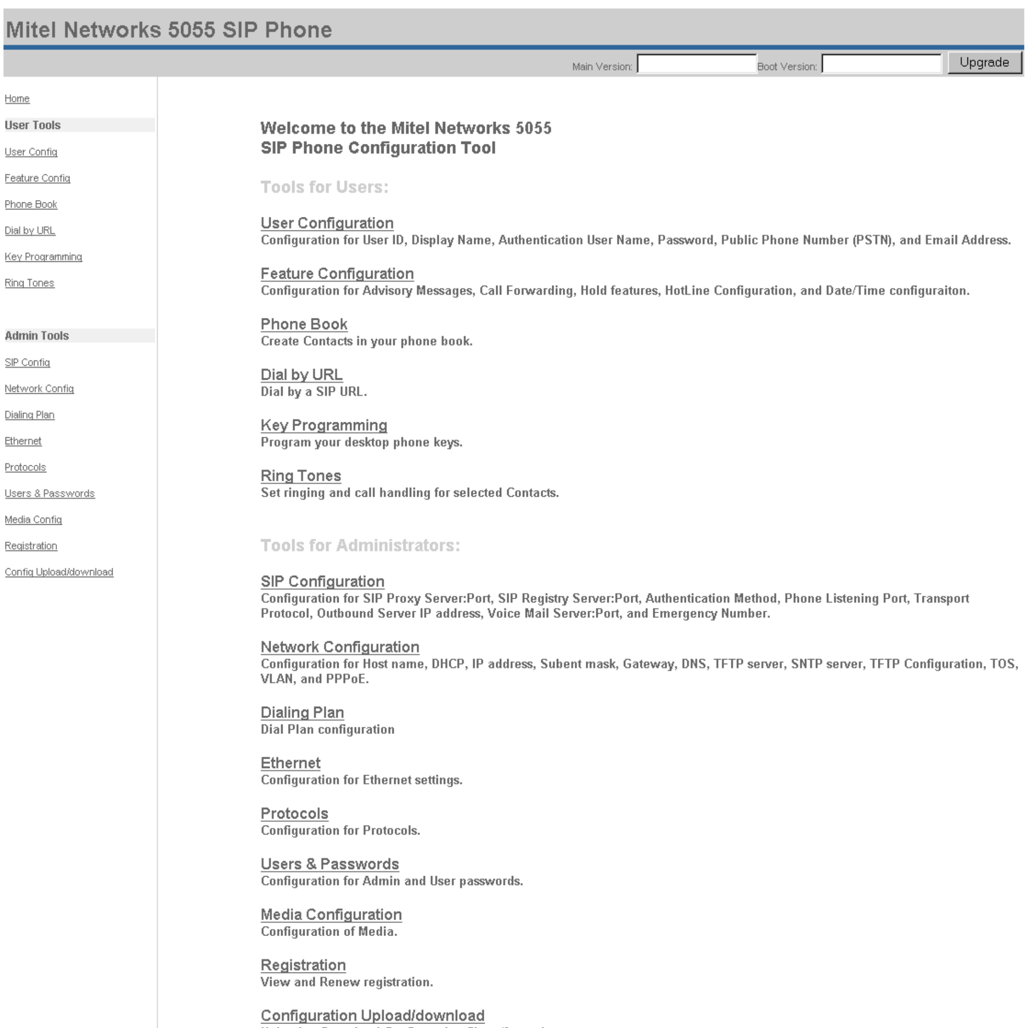
This appendix provides details on the settings and features available through the Web Configuration Tool.

Accessing the Web Configuration Tool

The Web Configuration Tool is accessed from any computer using a web browser. See *The Web Configuration Tool* on page 7 for instructions on accessing the Web Configuration Tool.

Home Page

Figure 6   Web Configuration Tool: Home Page



The Home Page shows the software version installed on the phone.

The **Upgrade** button displays the **Firmware Update** page.

User Configuration Page

The User Configuration page lets you change your user profile's basic parameters. After making the changes, click the **Save and Reboot** button; this will reboot your phone.

Figure 7 Web Configuration Tool: User Configuration Page

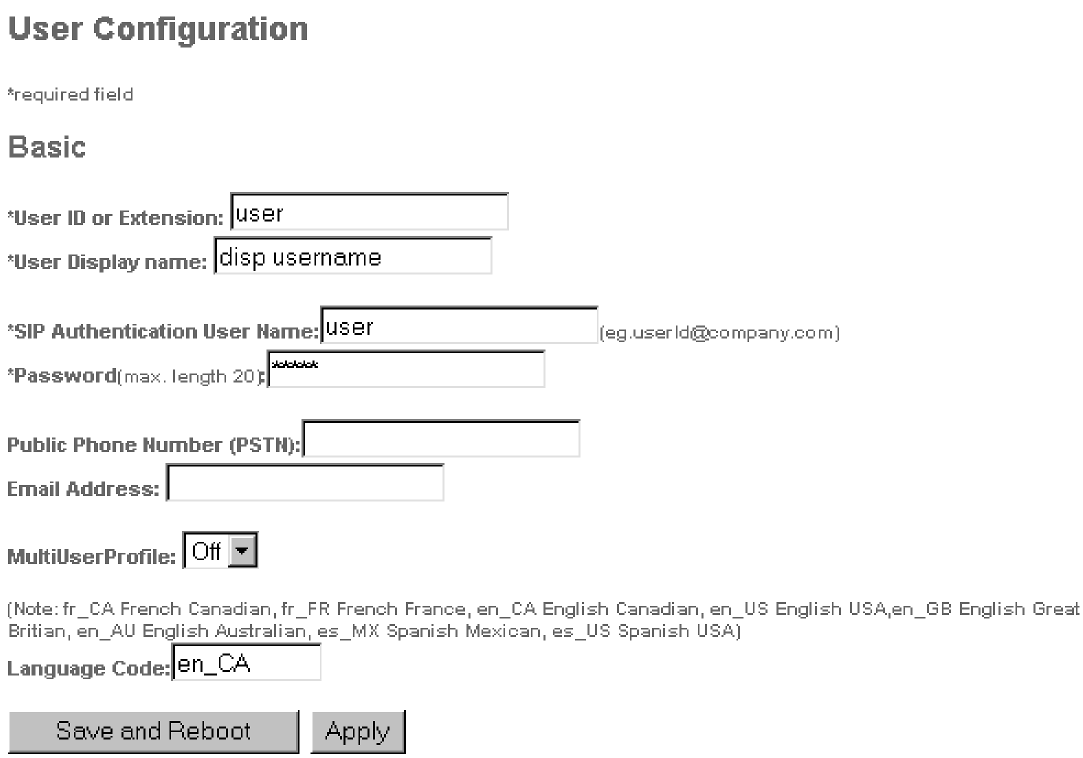


Table 5 Web Configuration Tool: User Configuration Settings

Setting Name	Values (bold = default)	Notes
Basic		
User ID or Extension	<text>	Unique name or number assigned to you. Limit of 32 characters. Default is **user** for default user profile.
User Display name	<text>	The name displayed on your phone when your user profile is active. Limit of 20 characters. Default is **display username** for default user profile.
SIP Authentication User Name	<text>	Your SIP Account name provided by your SIP Service Provider. Required only if you have a SIP Service Provider. Default is **user** for default user profile.

Setting Name	Values (bold = default)	Notes
Password	<text>	Your SIP account password provided by your SIP Service Provider. If you do not have a SIP Service Provider, password used to log in and activate your user profile.
Public Phone Number (PSTN)	<numbers>	Public phone number used as a possible alternative contact. Default is **blank**.
Email Address	<text>	E-mail address used as a possible alternative contact. Default is **blank**.
MultiUser Profile	**off**	MultiUser profiles work best with SIP phones that are being used in stand-alone installations (not directly connected to a 3050 ICP). Turn this feature on, only if your SIP phone is stand-alone, and you need this feature.
Language Code	 *en_CA* *fr_CA* *fr_FR* *en_US* *en_GB* *en_AU* *es_MX* *es_US*	Changes the phone display language English Canadian French Canadian French France English USA English Great Britain English Australian Spanish Mexican Spanish USA

Feature Configuration Page

The Feature Configuration page lets you program a number of settings attached to your user profile. After making the changes, click the **Apply** button.

Figure 8 Web Configuration Tool: Feature Configuration Page

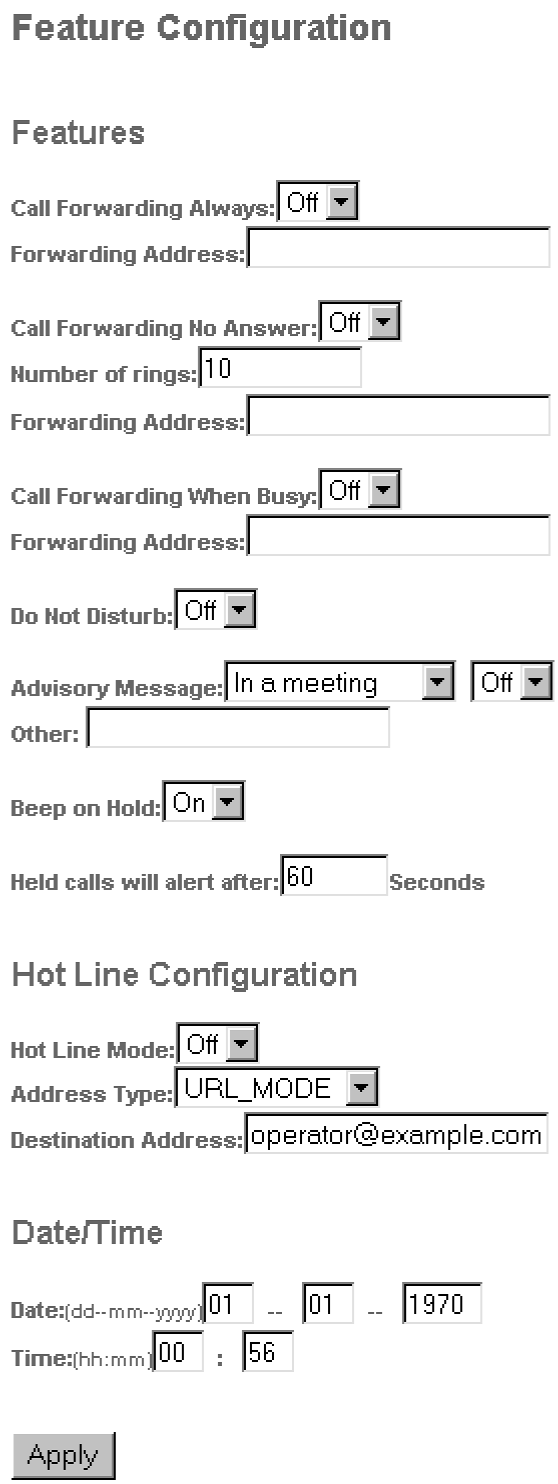


Table 6 Web Configuration Tool: Feature Configuration Settings

Setting Name	Values (bold = default)	Notes
Features		
Call Forwarding Always	On | **Off**	
Forwarding Address	SIP address.	Default is **blank**. If left blank calls will be forwarded to voice mailbox as programmed in SIP configuration page.
Call Forwarding No Answer	On | **Off**	
Number of Rings	< number>	Default value is **10**.
Forwarding Address	SIP address.	Default is **blank**. If left blank calls will be forwarded to voice mailbox as programmed in SIP configuration page.
Call Forwarding When Busy	On | **Off**	
Forwarding Address	SIP address.	Default is **blank**. If left blank calls will be forwarded to voice mailbox as programmed in SIP configuration page.
Do Not Disturb	On | **Off**	
Advisory Message	On | **Off**	
Message	**In a Meeting** | Out of town, At lunch | On vacation | In a Conference | Back in 5 minutes | Gone Home | Off Sick | Other reason	To enter a personalized message, select “Other reason”.
Other Reason	<message>	Fill in if “Other reason” is selected above. Limit of 20 characters.
Hold		
Beep on Hold	**On** | Off	Heard by user when on hold.
Held calls will ring back after:	<delay in seconds>	Heard by user to remind a call is on hold. Default value is **60** seconds
Hot Line Configuration		
Hot Line Mode	On | **Off**	When On, dials the Destination Address automatically when the SIP Phone goes off-hook.
Address Type	Num_Mode, **URL_Mode**	Num_Mode: number address. URL_Mode: URL address.
Destination Address	<address>	Must correspond to the type chosen in “Address Type” above. Default is **operator@example.com**.
Date/Time		
Date	<date with format day-month-year>	Modify only if there is no SNTP server (see Network Configuration page). These settings will be lost if the SIP Phone reboots.
Time	<time in 24-hour format>	

Phone Book Page

The Phone Book page lets you define up to five contacts for your user profile's Phone Book. You can also dial any of the contacts from this page. Click the **Apply** button after making any changes to your contacts. To dial a contact, select it in the drop down menu, and click the **Dial** button.

Figure 9 Web Configuration Tool: Phone Book Page

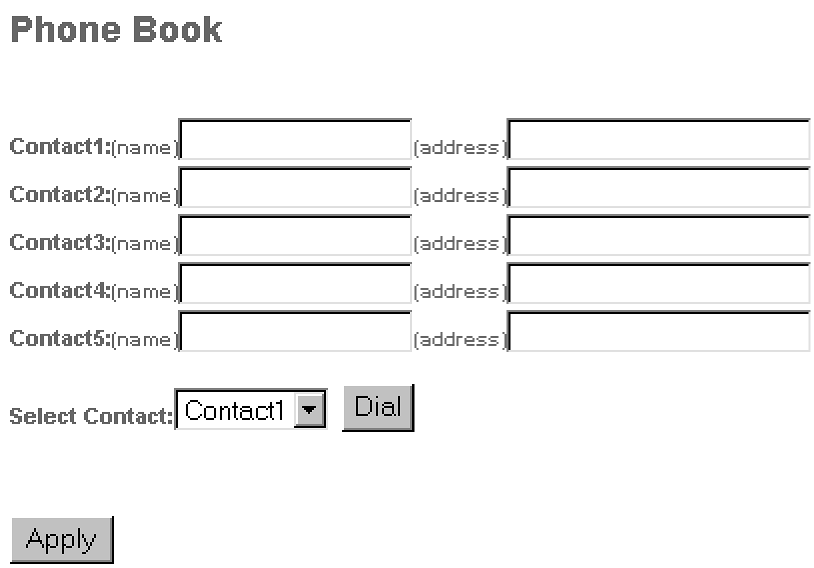


Table 7 Web Configuration Tool: Phone Book Settings

Setting Name	Values (bold = default)	Notes
Contact n		
Name	<name or nickname of the contact>	Limit of 20 characters.
Address	<number> | <name> | sip:<URL>	Limit of 128 characters.
Select Contact	**Contact 1** | Contact 2 | Contact 3 | Contact 4 | Contact 5	

Dial by URL Page

The Dial by URL page lets you dial a URL or IP address from the Web Configuration Tool. Click the **Dial** button to dial the URL. To save the SIP URL so that you need not re-enter it in the future, click the **Apply** button.

Figure 10 Web Configuration Tool: Dial by URL Page

Table 8 Web Configuration Tool: Dial by URL Settings

Setting Name	Values (bold = default)	Notes
Enter SIP URL	<number> | <name> | sip:<URL>	A URL must be preceded by “sip:”. Limit of 128 characters.

For examples of SIP URL syntax, refer to the table on the last page of this guide.

Key Programming Page

The Key Programming page lets you assign an address to one of the seven programmable speed-call keys. An address can be a name, number or URL.

Figure 11 Web Configuration Tool: Key Programming Page

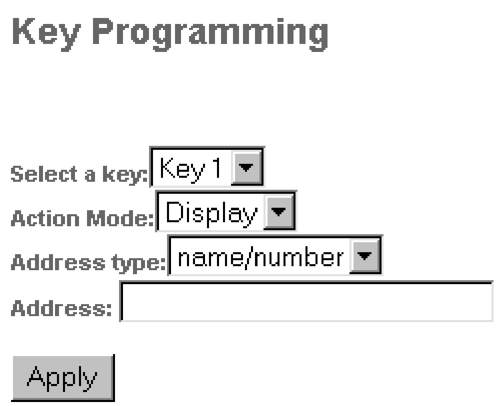


Table 9 Web Configuration Tool: Key Programming Settings

Setting Name	Notes
Select a key	Lets you select the key to be programmed (key 1, is the bottom-left key)
Action Mode	**Display** lists the key assignment. **Update** lets you change it.
Address Type	You must select the address type before entering it.
Address	Enter the address here.

Ring Tone Page

The Ring Tones configuration page lets you define up to three “rules” to control how the phone treats calls originating from individual callers (specified by SIP URL) or from groups of callers (specified by domain name). The phone can:

- Associate one of 12 different ring pitches to the call
- Automatically forward the call to voicemail
- Reject the call

Figure 12 Web Configuration Tool: Ring Tone Page

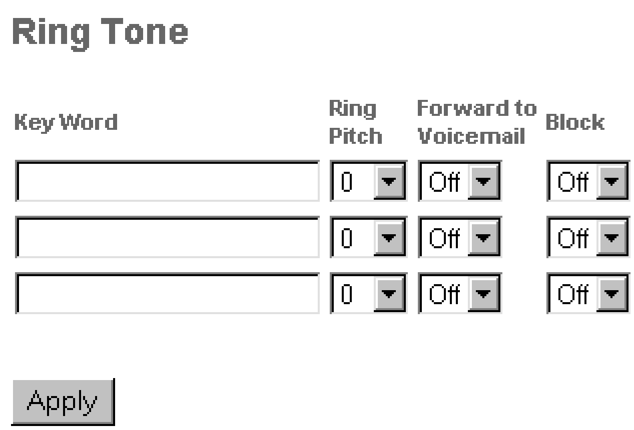


Table 10 Web Configuration Tool: Ring Tones Settings

Setting Name	Values (bold = default)	Notes
Key Word	sip:<URL> | domain name	A URL must be preceded by “sip:”. Domain names Limit of 128 characters.
Ring Pitch	**0** - 12	Selects the pitch that you will hear
Forward to Voicemail	**Off**/On	Forwards the caller automatically to your voicemail
Block	**Off**/On	Prevents the caller from reaching you.

SIP Configuration Page

The SIP Configuration page lets you change the SIP Service Provider configurations of your SIP Phone. These settings are specific to the SIP Phone. After making the changes, click the **Save and Reboot** button; this will reboot your phone.

Figure 13 Web Configuration Tool: SIP Configuration Page

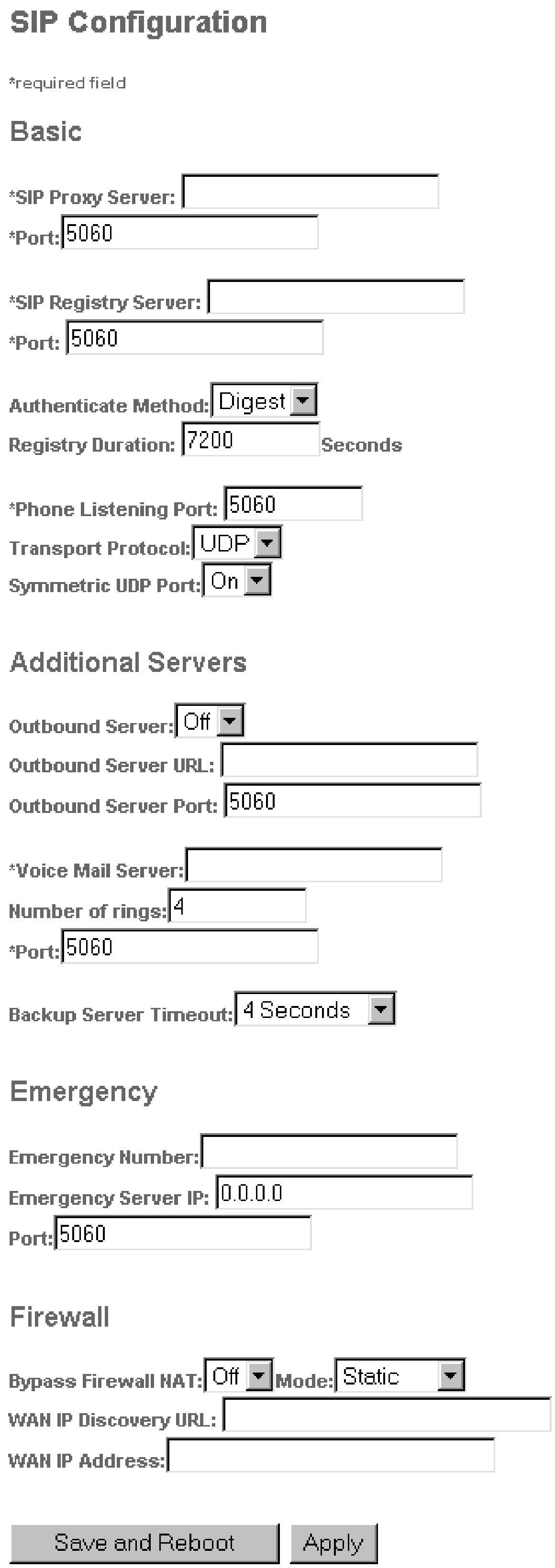


Table 11 Web Configuration Tool: SIP Configuration Settings

Setting Name	Values (bold = default)	Notes
Basic		
SIP Proxy Server	<IP address> | <domain name>	Appended to dialed name or number (for example, name@proxy). Limit of 128 characters. Default is **blank**.
Port	<number>	The SIP Proxy Server port number. Default is **5060**.
SIP Registry Server	<IP address> | <domain name>	Used if SIP Proxy and Registry Servers are not the same. Limit of 128 characters. Default is **blank**.
Port	<number>	The SIP Registry Server port number. Default is **5060**.
Authenticate Method	None | **Basic** | Digest	None: no registration authentication. Basic: authentication without encryption. Digest: authentication with encryption.
Registry Duration	<duration in seconds>	Time after which you are automatically deregistered. Default value is **7200** seconds (2 hours).
Phone Listening Port	<number>	Receive port used by the SIP Phone for SIP signaling. Default is **5060**.
Transport Protocol	**UDP** | TCP	Default type of packets for transmitted SIP signaling. UDP = User Datagram Protocol. TCP = Transmission Control Protocol.
Symmetric UDP Port	**symmetric**	**Symmetric is recommended**
Additional Servers		
Outbound Server	On | **Off**	If on, all SIP request and responses are sent to the outbound server. Default is **192.168.0.1**.
Outbound Server URL	<blank>	
Outbound Server Port	<blank>	
Voice Mail Server	<IP address> | <domain name>	Server address of external voice mail server. Default is **blank**. If this field is configured, the phone will connect to server using the default user name during boot-up.
Number of rings	**4**	
Port	<number>	Port number of Voice Mail Server. Default value is **5060**.
Backup Server Timeout	**4**	Some ISPs offer backup servers that can be used when the primary server is unavailable. This field lets you configure how long the 5055 SIP phone will wait before it tries the backup server.
Emergency		
Emergency Number	<number>	Emergency number for area, if applicable (for example, most of North America uses 911 as an emergency number). Default is **blank**.
Emergency Server IP	<IP address>	Address of server used by emergency number dialed. Default is **0.0.0.0**.
Port	<number>	Port number of Emergency Server. Default is **5060**.

Setting Name	Values (bold = default)	Notes
Firewall		
Bypass Firewall NAT	On | **Off**	This enables or disables firewall NAT bypass operation. When enabled, this features lets the 5055 SIP phone function behind a firewall which is not SIP-aware.
Mode	static | **dynamic**	Determines how the 5055 SIP phone will obtain the IP address of the firewall. A static address is
WAN IP Discovery URL	<URL>	WAN IP Discovery address
WAN IP Address	<IP address>	Static WAN IP Address

Network Configuration Page

The Network Configuration page lets you change the network configurations of your SIP Phone. These settings are specific to the SIP Phone. After making the changes, click the **Save and Reboot** button; this will reboot your phone.

Figure 14 Web Configuration Page: Network Configuration Page

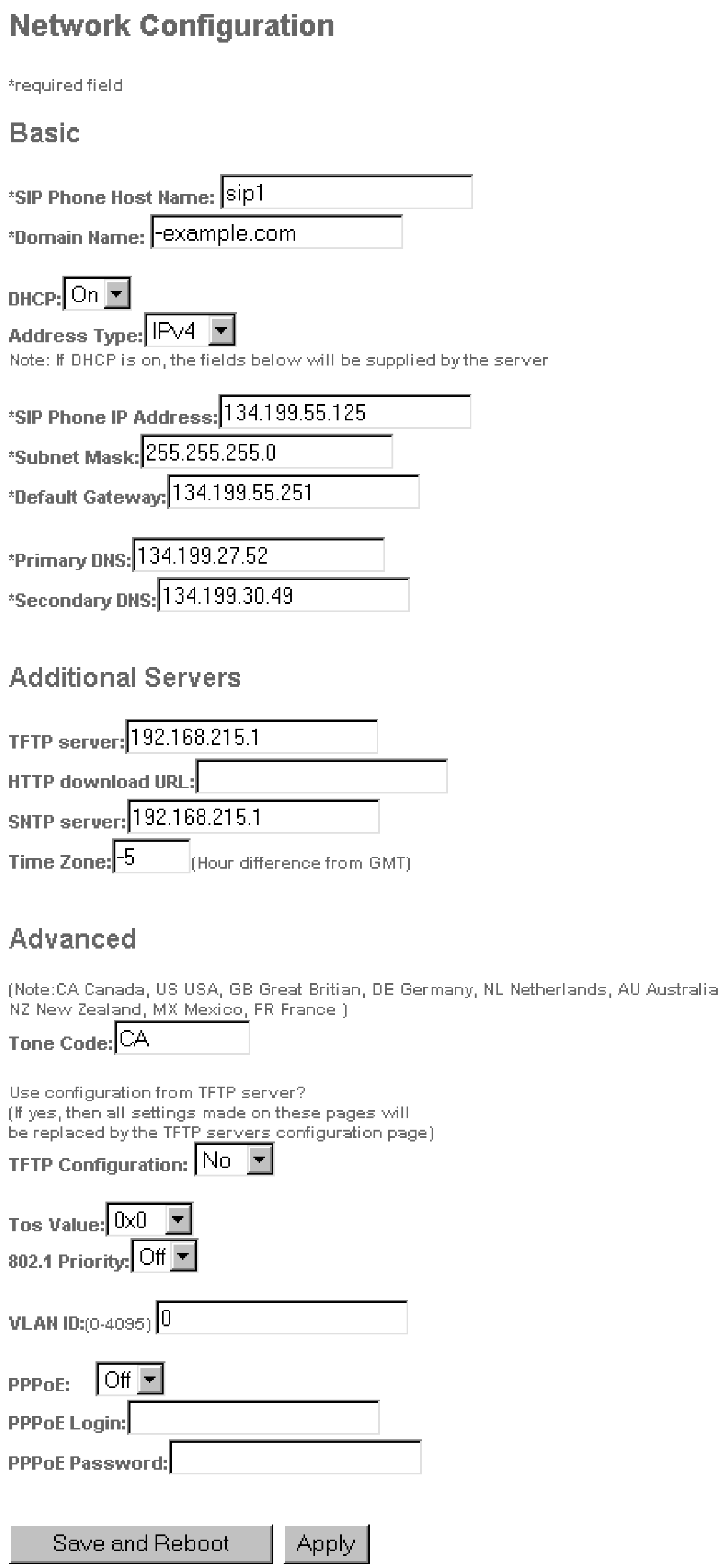


Table 12 Web Configuration Tool: Network Configuration Settings

Setting Name	Values (bold = default)	Notes
Basic		
SIP Phone Host Name	<host name>	Required for cable access. Default is **sip1**.
Domain Name	<domain name>	Optional. Limit of 128 characters. Default is **–example.com**.
DHCP	**On** | Off	If On, allows your ISP or LAN to allocate you a dynamic IP address.
Address Type	**IPv4** | Fqdn	IPv4: outgoing SIP requests use dotted format of IP address. Fqdn: outgoing SIP requests use “sip:host_name.domain” format for “contact” SIP header.
SIP Phone IP Address	<IP address>	Required. Provided automatically by ISP or LAN when DHCP is On[1].
Subnet Mask	<IP address>	Required. Provided automatically by ISP or LAN when DHCP is On[1]. Default is **255.255.255.0**.
Default Gateway	<IP address>	Required. Provided automatically by ISP or LAN when DHCP is On[1].
Primary DNS	<IP address>	Required. Provided automatically by ISP or LAN when DHCP is On[1].
Secondary DNS	<IP address>	Optional. Provided automatically by ISP or LAN when DHCP is On[1].
Additional Servers		
TFTP Server	`sipdnld.mitel.com`	Optional. The server where updates to the firmware and languages can be downloaded using the TFTP protocol.
HTTP download URL	<URL>	Optional. The address of the HTTP server where updates to the firmware and languages can be downloaded. HTTP is an alternative protocol to TFTP. If you leave this filed blank, the phone will attempt to use TFTP to update software. If HTTP update fails, then the phone does not automatically “fall back” to try TFTP.
SNTP Server	<IP address>	Optional. Server used for date/time synchronization. Default is **192.53.103.103**.
Time Zone	<number>	Optional. Difference between GMT and local time. Default is **–5** (Eastern Standard Time)

[1] If you change this value manually with DHCP On, it will be overwritten by the ISP/LAN the next time the phone is rebooted.

Setting Name	Values (bold = default)	Notes
Advanced		
Tone Code	***CA*** *US* *GB* *DE* *NL* *AU* *NZ* *MX* *FR*	Changes the phone tone plan. Canada USA Great Britain Germany Netherlands Australia New Zealand Mexico France
TFTP Configuration	Yes | **No**	If set to yes, the SIP Phone looks for configuration files on the TFTP server when it boots up. The settings in the configuration files will overwrite any manually entered settings. Default is **192.168.0.1**.
ToS value	**0** - 1e (even numbers only)	Type of Service. QOS parameter used to define packet priority for the IP layer. You can select values from the pull-down menu in hexadecimal notation.
802.1 Priority	**Off** | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7	QOS parameter used to define packet priority for Ethernet layer.
VLAN ID	<0–4095>	Virtual LAN Id. Used by network administrators. Default is **1**.
PPPoE	On | Off	Point-to-Point Protocol over Ethernet. Enabled when using a DSL network.
PPPoE Login	<user/login name>	Required for DSL. Provided by DSL ISP.
PPPoE Password	<password>	Required for DSL. Provided by DSL ISP.

Time Zones

The table below is provided for your convenience. Mitel Network does not guarantee its accuracy. For Daylight Savings Time, add +1 (for example, if your standard time is –5, your daylight savings time is –4).

Table 13 Major Time Zones

Time Zone	Diff. from GMT
Fiji, New Zealand, Marshall Islands (US)	±12
Samoa, Midway Islands	-11
Cook Island, Hawaiian Standard Time (US)	-10
Alaska Standard Time (US)	-9
Pacific Standard Time (US/Canada)	-8
Mountain Standard Time (US/Canada)	-7
Central Standard Time (US/Canada), Mexico, Central America	-6
Eastern Standard Time (US/Canada), Caribbean, Colombia, Ecuador, Peru	-5
Atlantic Standard Time (US/Canada), Dominican Republic, Bolivia, Paraguay, Chile, Venezuela	-4
Newfoundland (Canada)	-3.5

Time Zone	Diff. from GMT
Mid-Atlantic	-2
Azores, Cape Verde Islands	-1
GMT, Western Europe Time, Morocco, Mali, Burkina Faso, Togo	0
Central Europe Time, Algeria, Chad, Angola	+1
Eastern Europe Time, Libya, Sudan, Mozambique, South Africa, Russia (East)	+2
Moscow Time, Ethiopia , Tanzania, Madagascar	+3
Iran	+3.5
Russia Center, Armenia, Georgia, Oman, United Arab Republic	+4
Afghanistan	+4.5
Pakistan, Turkmenistan	+5
India, Nepal, Sri Lanka	+5.5
Bangladesh, Bhutan, Tajikistan, Kazakhstan	+6
Myanmar	+6.5
Cambodia, Laos, Thailand, Vietnam	+7
Australian Western Standard Time, China, Indonesia, Mongolia, Philippines	+8
Korea, Japan,	+9
Australian Center Standard Time	+9.5
Australian Eastern Standard Time, Papua New Guinea	+10
Russia (East), Solomon Islands, Vanuatu	+11

Dialing Plan

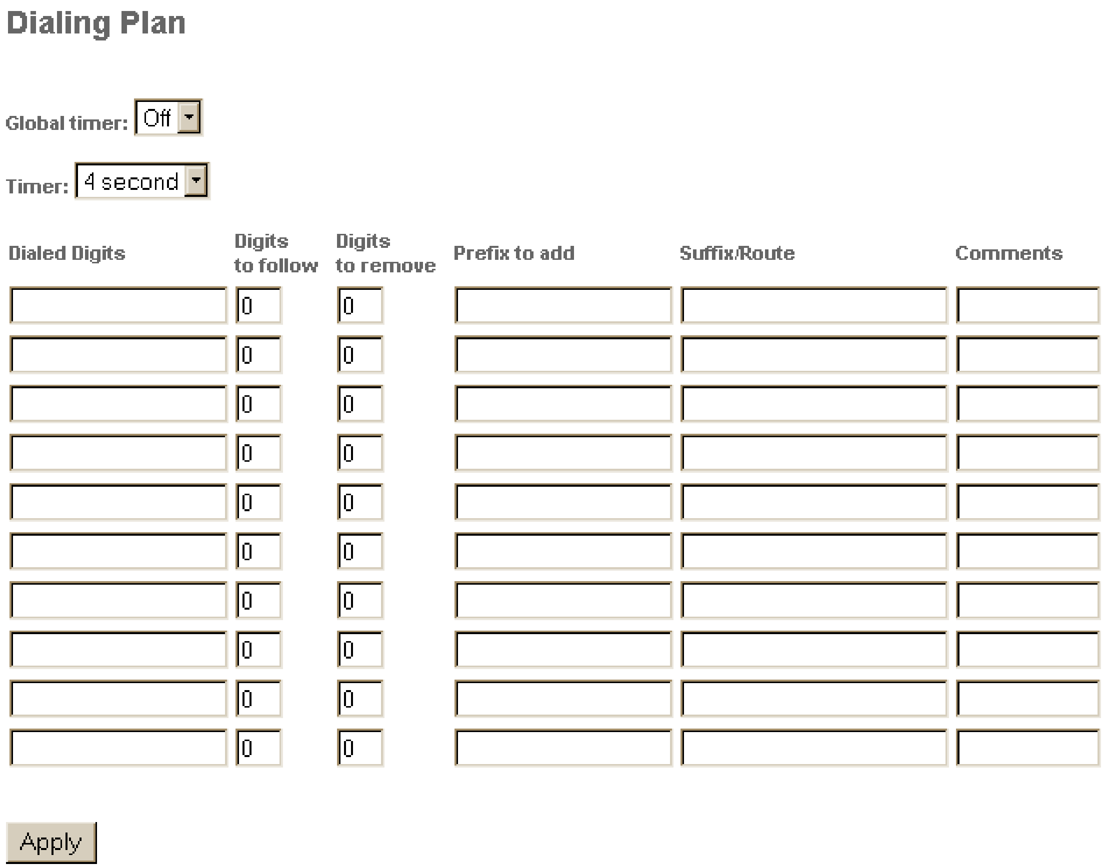
The Dialing Plan table lets you define Dialed Digit matching patterns that are used to test digits as they are entered. When a match is found, optional procedures such as removing leading digits, adding prefix or suffix digits may be performed before the resultant string of digits is sent to your SIP registration server, or a possible alternate server for dialing.

Effective use of dial plan rules can eliminate the need for you to terminate many common calls by pressing the **Dial** soft-key, and can simplify calls to special services such as discount long distance carriers.

The SIP phone tests digit strings using the top-most row first, then if no match is found here, it proceeds down the table until either a match is found, or all the rules are exhausted. If no match is found, the SIP phone will send the digits exactly as they were dialed to your SIP registration server for dialing.

Whether or not you have to press the **Dial** soft-key before the digits are sent, is dependent on the setting of the Global timer parameter

Figure 15   Dialing Plan Page



Note: Due to memory limitation in the phone, the contents of all Dialing Plan fields cannot exceed a total of 120 characters. If you exceed this limit, new characters will not be saved (you will notice this the next time you display the Dialing Plan table). Blank fields consume two "characters".

Table 14 Web Configuration Tool: Dialing Plan

Setting Name	Values (bold = default)	Notes
Global timer	Off | On	When activated, this optional feature forces all dialed digits to use the inter-digit timer specified in **Timer**. The **Dial** soft-key will be disabled as the dialed digits will be dialed automatically after the timer has expired.
Timer	1 | 2 | 3 | 4| 5 | 6 | 7 | 8 | 9 seconds	Optional parameters sets the duration of the inter-digit timer. When invoked using the ".T" parameter or the Global timer setting, the timer monitors the keypad for a pause in digit entry. When a pause is detected, the digits are optionally modified, then sent to the SIP registration server or alternate server.
Dialed Digits	<blank>	Enter a partial or complete digit string (of a minimum 2-digit length) for use as a template to test dialed digits. Matching digit strings are optionally manipulated, then sent to a SIP server for dialing If the value of one or more digits is likely to be variable, then you can enter the wildcard character "x" in place of each variable digit. The square brackets [ ] can be used to specify a set of digits, any one of which can match a dialed digit. For example [123] would yield a positive match if the user dials "1", "2" or "3", but would fail to match "4". The ".T" timer parameter (which must be placed at the end of a digit string) causes the phone to accept an arbitrary number of digits form the user. Once the user has finished entering digits (signaled by no new digits for a period equal to the timer value), the digit string: • may have digit manipulation performed on it • is sent to the SIP registration server without requiring the user to press the **Dial** soft key. **Note**: The digits to follow field must be set to 0 for the .T parameter to work. Maximum 16 digits.
Digits to follow	0	Optionally, enter the number of digits expected to follow the partial number specified under Dialed Digits here.
Digits to remove	0	Optionally, enter the number of leading dialed digits to be deleted from the dialed number here.

Table 14 Web Configuration Tool: Dialing Plan Continued

Setting Name	Values (bold = default)	Notes
Prefix to add	<blank>	Optionally, enter prefix digits here. Telephony digits 0 through 9 are valid. For DTMF trunks, 0 through 9, are valid. Maximum 16 digits.
Suffix/Route	<blank>	Can be used to specify suffix digits that are added to the digit string before it's sent to the SIP registration server for dialing. Alternatively, this field can be used to specify the URL of another SIP registration server, (an alternate route) If left blank, the digits will be routed to the default SIP registration server. Maximum 64 alphanumeric characters
Comments	<blank>	Use this optional field to label a digit plan entry. Maximum 8 characters.

Figure 15 Example Dial Plan Rules

1) Dialed 4-digit extensions beginning with the digits 3, 4 or 5 will be matched by this rule which makes use of wildcards to accommodate any three digits. The use of the Global time with this rule is optional.

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
[345]xxx	0	0			

2) Dialed 7-digit (local calls in North America) numbers will be matched by this rule which strips off the leading "9" before sending it to the SIP server for dialing. The use of the Global time with this rule is optional.

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
9[23456789]	0	1			Lcl Call

3) Dialed 10-digit (long distance in North America) numbers will be matched by this rule which strips off the leading "1" before sending it to the SIP server for dialing. The use of the Global time with this rule is optional.

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
91	10	1			Lng Dist

4) This example illustrates the use of a Timer triggered by the Dialed Digit pattern "9011" (long distance overseas call), with an arbitrary number of digits following it. The Global Timer should be off in this case since the Timer parameter is used.

Dialed Digits	Digits to follow	Digits to Remove	Prefix to Add	Suffix/Route	Comments
9011.T	0	1			Lng Dist

Ethernet Page

For two Ethernet devices to connect, they must share the same speed setting. Most current devices are capable of negotiating compatible settings automatically, as soon as you connect them. Some older equipment does not support auto negotiation, for this equipment, you will have to use this page to select the Ethernet operating parameters manually

Figure 16 Web Configuration Tool: Ethernet Page

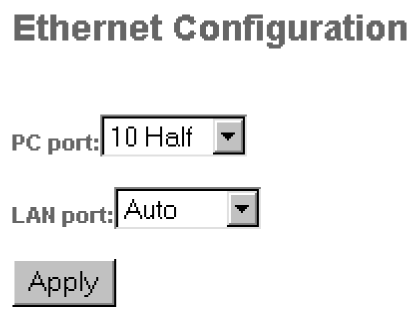


Table 16 Web Configuration Tool: Ethernet Settings

Setting Name	Values (bold = default)	Notes
PC port	**Auto** 10 Half 10 Full 100 Half 100 Full	The Auto setting lets the 5055 SIP phone and the PC negotiate the Ethernet speed and duplex automatically. If the PC does not support auto-negotiation, you must select a compatible setting manually.
LAN port	**Auto** 10 Half 10 Full 100 Half 100 Full	The Auto setting lets the 5055 SIP phone and the Ethernet device (hub, router, layer 2 switch, broadband modem etc.) negotiate the Ethernet speed and duplex. If the device does not support auto-negotiation, you must select a compatible setting manually.

Protocols Page

The Protocols Configuration page lets you activate and deactivate protocol support, if required for security reasons.

Figure 17 Web Configuration Tool: Protocols Page

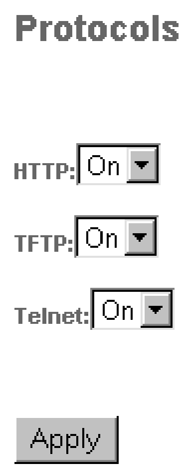


Table 17 Web Configuration Tool: Protocols Page

Setting Name	Values (bold = default)	Notes
HTTP	**On**, Off	Deactivating this protocol prevents future web-based sessions with the phone. To reactivate this protocol, you will have to use the SIP Phone interface.
TFTP	**On**, Off	
Telnet	**On**, Off	

Users & Passwords Page

The Security Configuration Page lets the administrator change the passwords for the user profiles and the administrator. This page can only be accessed using the Administrator user name and password. After making the changes, click the **Apply** button.

Figure 18 Web Configuration Tool: Users & Passwords Page

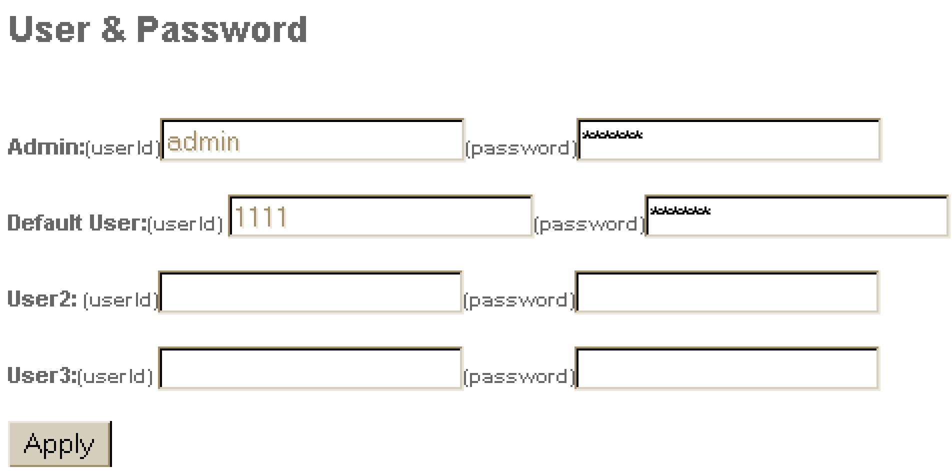


Table 18 Web Configuration Tool: Users & Passwords Settings

Setting Name	Values (bold = default)	Notes
User ID	As programmed.	Cannot change this value here. Defaults are **admin** for Administrator, and **user** for default user profile.
Password	Change as needed.	Default is **5055** for Administrator.

Media Configuration Page

The Media Configuration page lets you change the media configurations of your SIP Phone. These settings are specific to the SIP Phone. After making the changes, click the **Apply** button.

Figure 19 Web Configuration Tool: Media Configuration Page

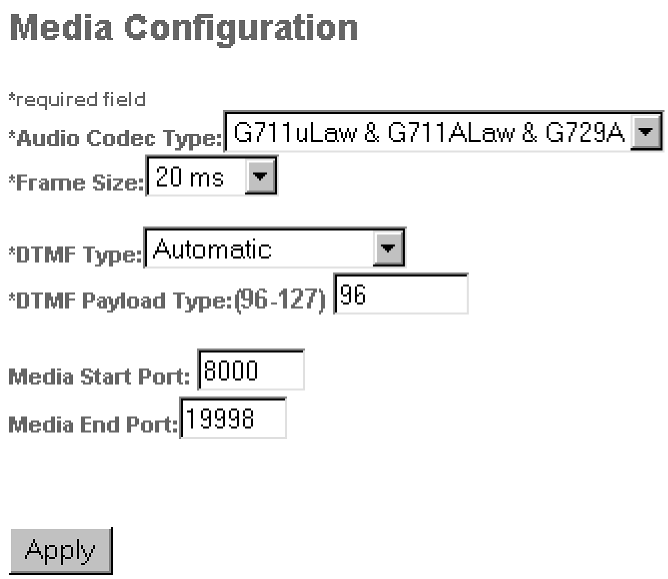


Table 19 Web Configuration Tool: Media Configuration Settings

Setting Name	Values (bold = default)	Notes
Basic		
Audio Codec Type	**G711uLaw** G711Alaw G729A G729A & G711uLaw G729A & G711uLaw & G711ALaw	G711 µ-Law: used in North America. G711 A-Law: used in the U.K. G729A: 8:1 compression codec requiring both ends to support this standard (otherwise, reverts to G.711).[2]
Frame Size	10 ms **20 ms** 30 ms	Frame size used by G711 or G729 codecs for packet size. This can be set to 10, 20, 30 …. 100)
DTMF Type	**Automatic** Outband & Inband Outband Inband	Used for DTMF tone generation (Outband used for RTP DTMF).
DTMF Payload Type (96-127)	**96**	
Media Start Port	**8000**	These fields specify the UDP port range used by the 5066 SIP phone. Change this if the default range conflicts with other devices in your network.
Media End Port	**19998**	

[2] If you set the audio codec type to G.729, the users will not be able to use the Conference Call feature (the **Conf** softkey is replaced by **NA** to indicate the feature is not available).

Registration Page

The Registration Status page lets you:

- determine if your 5055 SIP phone is registered with a SIP server (and how long its been registered)
- manually initiate a registration request

Under normal use, registration is entirely automatic. This page is provided for troubleshooting purposes

Figure 20 Web Configuration Tool: Registration Page

Table 20 Web Configuration Tool: Registration

Setting Name	Notes
Registration Status Display	Indicates if the phone is registered and with which server.
Registration Status Control	Manually initiates a registration request.

Configuration Upload/Download Page

The Configuration Upload/Download page lets you save the phone configuration file on your PC, or load a previously stored phone configuration file into your 5055 SIP phone.

Figure 21 Web Configuration Tool: Configuration Upload/Download Page

Table 21 Web Configuration Tool: Configuration Upload/Download

Setting Name	Values (bold = default)	Notes
Configuration Upload	File name	Use the Browse Button to locate a saved configuration file.
Download		Saves a configuration file on disk

Note: The phone configuration restore command does not restore passwords. If you have changed you passwords from their system-default values, and saved the phone configuration, then restoring this configuration to either a different phone, or the original phone one after it’s had its factory default values restored, will not restore these passwords to their original state.

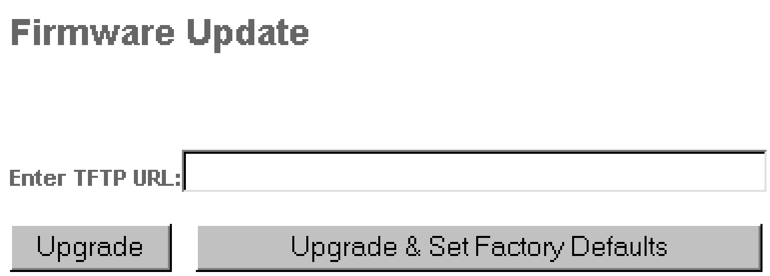
Upgrade

The Firmware Update page lets you enter a source URL for a software upgrade to the phone. The factory default configuration is URL: sipdnld.mitel.com which is the Mitel Networks firmware upgrade site.

Note: Though you can upgrade the phone's firmware using the manual key-method, it is preferable to use the **Upgrade** button on this page, as it saves and restores your current phone settings.

Clicking the "Upgrade" button preserves the current phone configuration, while upgrading the software (this is the preferred option). In some cases, the firmware release notes will instruct you to choose the second option "Upgrade & Set factory Defaults". This second option is required when differences between the software versions are so great, that the existing configuration cannot be preserved. When using the "Upgrade & Set factory Defaults" option, refer to the firmware release notes for instructions on how to restore you configuration from saved files.

Figure 22 Web Configuration Tool: Firmware Upgrade Page



Appendix C — Configuration Files

This appendix contains:

- Example of a Generic Configuration File
- Example of a Specific Configuration File
- Enabling Multiple User Profiles

Note: 1) The 5055 supports up to two HTTP Clients at the same time.

2) The CFG file must have the following as the first line:

image_name= NULL

3) Maximum CFG Upload capability to the phone is 3000 characters. The examples CFG files in this Appendix contain extensive comments. Working CFG files may not have room within the 3000 character limit to support detailed comments.

Example of a Generic Configuration File

```
# Mitel 5055 SIP Phone Generic Configuration File
# this file name=  SIPGeneric.cfg

# Grammar=
# comment lines are leading with a character '#'
# no escape '\' continue lines are allowed. No escape character is allowed.
# the string length of a parameter must be less than 128 characters
# parameter template=
# token = parameter ; comments

########################################################################
#             System Configuration Begins Here
########################################################################
# image version
image_name= NULL

# configure the phone by the tftpserver
# choice [0-no, 1- yes, 2-always]
# if the option "always" is being chosen, every time the phone boots up, it will download

#    the configuration parameters from the TFTP server which will overwrite any static
#    values of these parameters. This mode is useful for administrators to control the
#    phone's settings. User can't select this option from the web interface.
# If the option "Yes" is being chosen, the phone will boot up and download the
configuration
#    file from the TFTP server. Therefore, the statically configured parameters, if any,
#    will be overwritten by the parameters in the configuration file. After boot up, user
can
#    change this parameter to "No" from web interface.
# If the option "No" is being chosen, the phone, when booting up, will
#    not download the configuration file from the tftp server. The phone may only
#    do a version check for the main image.
tftp_config= 1

# address type
# choice[0-IPv4 or 2-FQDN],
# when the option IPv4 is being chosen, the outgoing sip requests will use the dotted
#    format of the IP address
# when the choice is FQDN, the outgoing sip requests will use the "sip:host_name.domain"
format
```

```
#    for the "contact" sip header. The FQDN address must be a resolvable entry in the DNS
server.
addr_type= 0

# domain name
domain= -example.com; domain name

# tftp server ip address
# the sip phone will download the boot image, the main image, and the configuration
parameters
# from this ip address
tftp= 192.168.0.1

# sntp server URL address
# the sip phone will update its date and time from this server
sntp= ntp.cpsc.ucalgary.ca

# time zone
# integer [-12, 0, 12]
time_zone= -5;  EST time
#time_zone= 8;  China

# tos
# integer [0, 1e] (even numbers only);
tos= 0

# IEEE 802.1 priority
# integer [-1, 0, 7]
# -1 means OFF
802_priority= -1

# VLAN ID
# integer[0, 4095]
vlan_id= 0


###################################################################################
# SIP configuration
###################################################################################

# sip phone will listen for the SIP packets at this port
# when the other phone calls this phone, the sip packets must be sent to this port
local_sip_port= 5060

# transport protocol for sip
# this parameter can be overwriten by dialing a url with this parameter: transport= udp
#    or transport = tcp
# choice [1-tcp, 2- udp]
trans_protocol= 2

# sip proxy address
# When dialing a number or user_id only, the proxy address will automatically be appended

# to the number (or id) as number@proxy or user_id@proxy
# The proxy address can be in IP address or domain.com format
proxy_addr=

# sip proxy port
proxy_port= 5060

# outbound_state
# choice [0-NO, 1-YES]
# if YES, all the sip requests and responses will always be sent to the outbound_ip.
# Otherwise, the sip responses will be sent to the "via" address. The requests will be
sent to
# "route" or "contact", according to the rule defined by the sip specs.
outbound_state= 0

# outbound proxy ip address
# 1) if, for some reason, the sip phone must send request to a local sip proxy first,
#    config this outbound sip proxy ip address
```

```
# 2) if the above proxy_addr (domain.com) is not resolvable from the DNS server,
#    this ip address will be used in place of the proxy address
outbound_ip= 192.168.0.1

#outbound server proxy port
outbound_port= 5060



# sip registrar address
# could be same as or different from the proxy_addr above
registrar=

# sip registrar port
registrar_port= 5060

# Registration duration in seconds for each register request
# the server may respond with a different duration
register_expire= 7200; in seconds

# Registration authentication method
# choice [0-NONE, 1-BASIC, 2-DIGEST]
auth_method= 0

# sip voice mail server addresssip phone willsend the "subsrib" request for message-
summary to this address
voice_mail_srv=

# auto forward to voice mail server after num of rings
voicemail_ringnum=4

# sip voice mail server port
voice_srv_port= 5060

# emergency number
# integer string
# when user dials this string, the phone will send the sip request to e911_ip
emerg_number= 911

# emergency ip address
# must be an ip address
emerg_ip= 192.168.0.1

# e911 port number
emerg_port= 5060

##################################################################################
# Media configuration
##################################################################################

# audio codec to offer
# the codec(s) you choose here, will be listed in the INVITE or OPTION's SDP
# choice 0-g711 uLaw
#        1-g711 ALaw
#        2-g729A
#        3-g729A and g711 uLaw
#        4-all of the above codecs
audio_codec=0

# audio codec packet size
# currently this parameter is only applicable to the g711 codec
# choice [10, 20, 30] ms
audio_pkt_size= 20;  ms

# dtmf type
# defines the way the DTMF digits will be sent across
# choice [0-automatic, 1-outband & inband, 2-outband only, 3-inband only
# automatic means when a "telephone-event" is being received from the peer party, send
the
# DTMF digit in the outband-only mode
#
```

```
dtmf_type= 0

#dtmf payload(96-127)
dtmf_payload= 96

#Media port configuration start port
start_port= 8000

#Media port configuration start port
end_port= 19998

##################################################################################
# feature configuration
##################################################################################

# auto answer mode
# choice [0-disabled, 1-enabled]
auto_answer= 0

# auto answer reason code
#         0- in a meeting
#         1- out of town
#         2- at lunch
#         3- on vacation
#         4- in a conference
#         5- in lab
#         6- back in 5 minutes
#         7- gone home
#         8- on a course
#         9- off sick
#         10- other reason
reasons= 0

# other reason string
# when reasons = 10, the sip will copy this string to the "reject reason" field.
other_reason= i am busy now!

# do not disturb
# choice [0 - disable, 1-enable]
# when enabled, all incoming calls will be rejected, or forwarded to the voicemail
do_not_disturb= 0

# call forwarding no answer mode
# choice [0-disabled, 1- enabled]
# when enabled, the incoming call will be forwarded to the "noans_fwd_addr"
# the condition defined by the "try_ring_nums"
noans_fwd_mode= 0

# call forwarding no answer after defined number of rings
# integer [1, 20]
try_ring_nums= 4

# call forwarding no answer address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to the user_id@proxy
noans_fwd_addr= 1002

# call forwarding always mode
# choice [0-disabled, 1- enabled]
# when enabled, the incoming call will be forwarded to the "always_fwd_addr"
always_fwd_mode= 0

# call forwarding always address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to the user_id@proxy
always_fwd_addr= 1002

# call forwarding when busy mode
# choice [0-disabled, 1- enabled]
```

```
# when enabled, the incoming call will be forwarded to the "forward_addr"
busy_fwd_mode= 0

# call forwarding when busy forward address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to user_id@proxy
busy_fwd_addr= 1002

# beep on hold
# choice [0- disable, 1- enable]
# If enabled, the SIP phone, when being held by the peer party, will generate beeps in
the receiver.
# if the sip proxy or back-2-back UA supports music on hold, this feature should be
disabled
beep_on_hold= 1

# on hold ringback timer
# if the sip phone puts the peer party on hold, and the handset is being put down on the
cradle, the
# sip phone will play a ringback signal after a period defined by this parameter
# to alert the user that there is a call on hold.
on_hold_alert= 300; seconds


# hot line
# choice [0-disable, 1-enable]
# when enabled, whenever the user pickup the phone handset, the call is automatically
#   made to the hot_address.
hot_line= 0

# hot line address type
# choice [0-number or id mode, 1-sip url]
hot_addr_type= 1

# hot line address
# defines the address with the format defined by the hot_addr_type
hot_address= operator@-example.com


#adminId
adminId= admin

#admin password
admin_passwd=be6ad8761fe4eb9bb85934a2d21686bb

#admin displayname
admin_displayname=admin

#symetric SIP UDP
#choice[0-symetric SIP UDP, other-no-symetric SIP UDP]
#default 0
sym_udp= 0

#configuration change SIP notify
#choice[0-disabled,1-enabled] enable this only when phone is behind Mitel 3050 server
#default 0
ntfcfg= 0

#SIP backup server timeout period
#choice[2-2 seconds,3-4 seconds, 4-8 seconds,5-6 seconds]
#default 3
backupsvr_tout= 3
```

Example of a Specific Configuration File

```
# Mitel 5055 SIP Phone Configuration File
# this file name=  SXXXXXXXXXXXX.cfg
# where xxxxxxxxxxxx is the MAC address padded with 0s

# comment lines are leading with a character '#'
# no escape '\' continue lines are allowed
# the string length of the parameter must be less than 128 characters
# parameter template=
# token = parameter ; comments

########################################################################
#            Configuration Begins Here
########################################################################
#            System Configuration Begins Here
########################################################################
# image version
image_name= NULL

# configure the phone by the tftpserver
# choice [0-no, 1- yes, 2-always]
# if the option "always" is being chosen, everytime the phone boots up, it will download
#    the configuration parameters from the TFTP server which will overwrite any static
#    values of these parameters. This mode is useful for administrators to control the
#    phone's settings. User can't select this option from the web interface.
# If the option "Yes" is being chosen, the phone will boot up and download the
configuration
#    file from the TFTP server. Therefore, the statically configured parameters, if any,
#    will be overwriten by the parameters in the configuration file. After boot up, user
can
#    change this parameter to "No" from web interface.
# If the option "No" is being chosen, the phone, when booting up, will
#    not download the configuration file from the tftp server. The phone may only
#    do a version check for the main image.
tftp_config= 0

# address type
# choice[0-IPv4 or 2-FQDN],
# when the option IPv4 is being chosen, the outgoing sip requests will use the dotted
#    format of the IP address
# when the choice is FQDN, the outgoing sip requests will use the "sip:host_name.domain"
format
#    for the "contact" sip header. The FQDN address must be a resolvable entry in the DNS
server.
addr_type= 0

# host name
# defines the host name of the sip phone.
# This parameter is used when addr_type is set to FQDN
host_name= sip1


# domain name
domain= -example.com; domain name

# tftp server ip address
# the sip phone will download the boot image, the main image, and the configuration
parameters
# from this ip address
tftp= 192.168.0.1


# sntp server URL address
# the sip phone will update its date and time from this server
sntp= ntp.cpsc.ucalgary.ca

# time zone
# integer [-12, 0, 12]
```

```
time_zone= -5;  EST time
#time_zone= 8;  China

# tos
# integer [0, 1e] (even numbers only);
tos= 0

# IEEE 802.1 priority
# integer [-1, 0, 7]
# -1 means OFF
802_priority= -1

# VLAN ID
# integer[0, 4095]
vlan_id= 0

###################################################################################
# IP network configuration
###################################################################################
#dhcp (0 = disable, 1 = enable)
dhcpenable= 1

#ip address
ipadr= 192.168.0.1

#network mask
 ipmask= 255.255.255.0

#network gateway
ipgateway= 192.168.0.1

#primary dns server
ipdns= 192.168.0.1

#secondary dns server
ipscddns= 192.168.0.1

#pppoe(0 = disable, 1 = enable)
pppoe_enable= 0

#pppoe login
pppoe_login= NULL

#pppoe password
pppoe_passwd= NULL




###################################################################################
# user profile configuration
###################################################################################
#multiple user profile(0 = disable, 1 = enable)
multi_user_enable=0


# user id.
# used to register to the sip proxy as user_id@domain
# this id must be one sting, no space is allowed
user_id= user

# user display name.
# used for display the user's human readable name in sip "from" header
#      from= "display name" sip= user_id@domain
disp_name= disp username

# user_name
# used as the user identify for authentication purpose. It could be same as user_id.
# but it could also be different. such as in the format of user_id@domain
```

```
# no white space is allowed in this string
user_name= user@-example.com

# password
# As a pair with user_name for authentication purpose.
password= hello

# User's other pstn phone number
# this parameter is used in SDP packets to show the user can also be reached by this
# phone number.
# this is an option.
phone_num=

# email address of the user
# this parameter is used in SDP packets to show the user can also be reached by this
# email address. It is an option
email= user_name@-example.com

###################################################################################
##   Additional users, up to 2 for release 1
###################################################################################
# user id.
# used to register to the sip proxy as user_id@domain
# this id must be one sting, no space is allowed
#user_id2= 1002

# user display name.
# used for display the user's human readable name in sip "from" header
#      from= "display name" sip= user_id@domain
#disp_name2= user 1002

# user_name
# used as the user identify for authentication purpose. It could be same as user_id.
# but it could also be different. such as in the format of user_id@domain
# no white space is allowed in this string
#user_name2= 1002@-example.com

# password
# As a pair with user_name for authentication purpose.
#password2= 1002

# user id.
# used to register to the sip proxy as user_id@domain
# this id must be one sting, no space is allowed
#user_id3= 1003

# user display name.
# used for display the user's human readable name in sip "from" header
#      from= "display name" sip= user_id@domain
#disp_name3= user 1003

# user_name
# used as the user identify for authentication purpose. It could be same as user_id.
# but it could also be different. such as in the format of user_id@domain
# no white space is allowed in this string
#user_name3= 1003@-example.com

# password
# As a pair with user_name for authentication purpose.
#password3= 1003


##################################################################################
# SIP configuration
##################################################################################

# sip phone will listen for the SIP packets at this port
# when the other phone calls this phone, the sip packets must be sent to this port
local_sip_port= 5060
```

```
# transport protocol for sip
# this parameter can be overwriten by dialing a url with this parameter: transport= udp
#    or transport = tcp
# choice [1-tcp, 2- udp]
trans_protocol= 2

# sip proxy address
# When dialing a number or user_id only, the proxy address will automatically be appended

# to the number (or id) as number@proxy or user_id@proxy
# The proxy address can be in IP address or domain.com format
proxy_addr=

# sip proxy port
proxy_port= 5060

# outbound_state
# choice [0-NO, 1-YES]
# if YES, all the sip requests and responses will always be sent to the outbound_ip.
# Otherwise, the sip responses will be sent to the "via" address. The requests will be
sent to
# "route" or "contact", according to the rule defined by the sip specs.
outbound_state= 0


# outbound proxy ip address
# 1) if, for some reason, the sip phone must send request to a local sip proxy first,
#    config this outbound sip proxy ip address
# 2) if the above proxy_addr (domain.com) is not resolvable from the DNS server,
#    this ip address will be used in place of the proxy address
outbound_ip= 192.168.0.1

#outbound server proxy port
outbound_port= 5060


# sip registrar address
# could be same as or different from the proxy_addr above
registrar=

# sip registrar port
registrar_port= 5060

# Registration duration in seconds for each register request
# the server may respond with a different duration
register_expire= 7200; in seconds

# Registration authentication method
# choice [0-NONE, 1-BASIC, 2-DIGEST]
auth_method= 0

# sip voice mail server addresssip phone willsend the "subsrib" request for message-
summary to this address
voice_mail_srv=

# auto forward to voice mail server after num of rings
voicemail_ringnum=4

# sip voice mail server port
voice_srv_port= 5060

# emergency number
# integer string
# when user dials this string, the phone will send the sip request to e911_ip
emerg_number= 911

# emergency ip address
# must be an ip address
emerg_ip= 192.168.0.1

# e911 port number
```

```
emerg_port= 5060

##################################################################################
# Media configuration
##################################################################################

# audio codec to offer
# the codec(s) you choose here, will be listed in the INVITE or OPTION's SDP
# choice 0-g711 uLaw
#        1-g711 ALaw
#        2-g729A
#        3-g729A and g711 uLaw
#        4-all of the above codecs
audio_codec=0

# audio codec packet size
# currently this parameter is only applicable to the g711 codec
# choice [10, 20, 30] ms
audio_pkt_size= 20;  ms

# dtmf type
# defines the way the DTMF digits will be sent across
# choice [0-automatic, 1-outband & inband, 2-outband only, 3-inband only
# automatic means when a "telephone-event" is being received from the peer party, send
the
# DTMF digit in the outband-only mode
#
dtmf_type= 0

#dtmf payload(96-127)
dtmf_payload= 96

#Media port configuration start port
start_port= 8000

#Media port configuration start port
end_port= 19998

##################################################################################
# feature configuration
##################################################################################
# auto answer mode
# choice [0-disabled, 1-enabled]
auto_answer= 0

# auto answer reason code
#         0- in a meeting
#         1- out of town
#         2- at lunch
#         3- on vacation
#         4- in a conference
#         5- in lab
#         6- back in 5 minutes
#         7- gone home
#         8- on a course
#         9- off sick
#         10- other reason
reasons= 0

# other reason string
# when reasons = 10, the sip will copy this string to the "reject reason" field.
other_reason= i am busy now!

# do not disturb
# choice [0 - disable, 1-enable]
# when enabled, all incoming calls will be rejected, or forwarded to the voicemail
do_not_disturb= 0

# call forwarding no answer mode
# choice [0-disabled, 1- enabled]
# when enabled, the incoming call will be forwarded to the "noans_fwd_addr"
```

```
# the condition defined by the "try_ring_nums"
noans_fwd_mode= 0

# call forwarding no answer after defined number of rings
# integer [1, 20]
try_ring_nums= 4

# call forwarding no answer address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to the user_id@proxy
noans_fwd_addr= 1002

# call forwarding always mode
# choice [0-disabled, 1- enabled]
# when enabled, the incoming call will be forwarded to the "always_fwd_addr"
always_fwd_mode= 0

# call forwarding always address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to the user_id@proxy
always_fwd_addr= 1002

# call forwarding when busy mode
# choice [0-disabled, 1- enabled]
# when enabled, the incoming call will be forwarded to the "forward_addr"
busy_fwd_mode= 0

# call forwarding when busy forward address
# sip url for forwarding the call
# in the format of user_id, or user_id@domain.com
# if only user_id is configured, the call will be forwarded to user_id@proxy
busy_fwd_addr= 1002

# beep on hold
# choice [0- disable, 1- enable]
# If enabled, the SIP phone, when being held by the peer party, will generate beeps in
the receiver.
# if the sip proxy or back-2-back UA supports music on hold, this feature should be
disabled
beep_on_hold= 1

# on hold ringback timer
# if the sip phone puts the peer party on hold, and the handset is being put down on the
cradle, the
# sip phone will play a ringback signal after a period defined by this parameter
# to alert the user that there is a call on hold.
on_hold_alert= 300; seconds


# hot line
# choice [0-disable, 1-enable]
# when enabled, whenever the user pickup the phone handset, the call is automatically
#   made to the hot_address.
hot_line= 0

# hot line address type
# choice [0-number or id mode, 1-sip url]
hot_addr_type= 1

# hot line address
# defines the address with the format defined by the hot_addr_type
hot_address= operator@-example.com


#adminId
adminId= admin

#admin password
admin_passwd=be6ad8761fe4eb9bb85934a2d21686bb
```

```
#admin displayname
admin_displayname=admin

#http protocol enable
# choice [0-disable, 1-enable]
http_task_enable= 1

#tftp protocol enable
# choice [0-disable, 1-enable]
tftp_task_enable= 1

#telnet protocol enable
# choice [0-disable, 1-enable]
telnet_task_enable= 1

#symetric SIP UDP
#choice[0-symetric SIP UDP, other-no-symetric SIP UDP]
#default 0
sym_udp= 0

#program key
pk1=
pk2=
pk3=
pk4=
pk5=
pk6=
pk7=

#configuration change SIP notify
#choice[0-disabled,1-enabled] enable this only when phone is behind Mitel 3050 server
#default 0
ntfcfg= 0

#SIP backup server timeout period
#choice[2-2 seconds,3-4 seconds, 4-8 seconds,5-6 seconds]
#default 3
backupsvr_tout= 3

#firmware upgrade http download url
http_download=

# language code
# fr_CA French Canadian, fr_FR French France, en_CA English Canadian, en_US English USA
# en_GB English Great Britian, en_AU English Australian, es_MX Spanish Mexican, es_US
Spanish USA
lancode= en_CA


#Ring tone code
# CA Canada, US USA, GB Great Britian, DE Germany, NL Netherlands, AU Australia, NZ New
Zealand,
# MX Mexico, FR France
tonecode= CA

#Dialing plan auto dialing global timer(0 disalbed, 1 enabled)
gtEnable= 0

#Dial plan auto dialing timer(1-9 secondes)
dtimer= 4

#dialing plan string( max len 256)
dialpl=

#firmware TFTP upgrade url
upgurl= sipdnld.mitel.com

###################################################################################
# fire wall configuration
###################################################################################
```

```
#firmware tranversal (0 disalbed, 1 enabled)
fwEnable= 0

#fire wall WAN address discovery mode(0 static, 1 dynamic)
fwMode= 0

#fire wall WAN address discovery url
fwWanDurl=

#fire wall WAN address
fwWanurl=


###################################################################################
# phone book configuration
###################################################################################
#phone book index(0-4)
pbIndex= 0

#phone book entry 1 name
pbName1=

#phone book entry 1 address
pbAddr1=

#phone book entry 2 name
pbName2=

#phone book entry 2 address
pbAddr2=

#phone book entry 3 name
pbName3=

#phone book entry 3 address
pbAddr3=

#phone book entry 4 name
pbName4=

#phone book entry 4 address
pbAddr4=

#phone book entry 5 name
pbName5=

#phone book entry 5 address
pbAddr5=

######################################################################################
# Distinctive Ring  configuration
#####################################################################################
#ring tone entery 1 key word
rdkw1=

#ring tone entry 1 type(0-16)
rdringtype1= 0

#ring tone entry 1 forward to voice mail(0 disalbed, 1 enabled)
rdvmail1= 0

#ring tone entry 1 block the call(0 diabled, 1 enabled)
rdblock1= 0

#ring tone entery 2 key word
rdkw2=

#ring tone entry 2 type(0-16)
rdringtype2= 0

#ring tone entry 2 forward to voice mail(0 disalbed, 1 enabled)
```

```
rdvmail2= 0

#ring tone entry 2 block the call(0 diabled, 1 enabled)
rdblock2= 0

#ring tone entry 3 key word
rdkw3=

#ring tone entry 3 type(0-16)
rdringtype3= 0

#ring tone entry 3 forward to voice mail(0 disalbed, 1 enabled)
rdvmail3= 0

#ring tone entry 3 block the call(0 diabled, 1 enabled)
rdblock3= 0


Note:   You can define the same parameters defined in  SIPGeneric.cfg here. When defined
        here, the parameters overwrite the values in the SIPGeneric.cfg file.
```

Enabling Multiple User Profiles

Use this procedure to enable Multi-User Profiles:

1. Navigate to "Configure Upload/Download Page"
2. Click on "Download" to save configuration parameters to download.txt file
3. Modify download.txt file entry from:

   `multi_user_enable= 0`

   to

   `multi_user_enable= 1`
4. From "Configure Upload/Download Page" browse to download.txt file and click on "Upload"

Appendix D — Working with Firewalls

The 5055 SIP phone can be configured to work behind Network Address Translation (NAT) firewalls which are not SIP aware by enabling the SIP configuration **Bypass Firewall NAT** feature and configuring the firewall correctly. To do this:

1. Locate the documentation that came with your NAT firewall and look for instructions on how to configure a Demilitarized Zone (DMZ) server. You must configure the 5055 SIP phone to function as a DMZ server to the firewall.

2. Use the 5055 SIP phone configuration web page to:

   A. Login to the phone using the web interface and select **Network Configuration** Configure a static IP address, Subnet Mask, Gateway, and DNS server address. Turn DHCP off.

   B. Select **SIP Configuration**. Set **Bypass Firewall NAT** On.

   C. In the "Mode" box select static if the IP of the WAN port of the router never changes. Select dynamic if the IP changes as the result of DHCP or PPPOE.

   D. If you selected static IP in step C., enter this IP in the **WAN IP Address** field. If you selected dynamic, enter the URL of the service provider in the "**WAN IP Discovery URL**" field.

   E. Click the **Apply** button and verify if **Bypass Firewall NAT** is still set to **On.** If dynamic IP address is used, also verify that **WAN IP Address field** has been filled in with a valid address. If **Bypass Firewall NAT** resets to **off,** or there is no IP address in the **WAN IP Address field**, then there is likely a problem with the **WAN IP Discovery URL** that is preventing the phone from obtaining the router's WAN IP address.

Tip: The 5055 SIP phone must be in a factory-default state (this configuration will not work with the phone registered or trying to register). To do this power-cycle the phone while holding down the 3 key and answer "yes" when asked if you want to "use factory default."

Tip: There are some service providers that provide free Dynamic IP services. If your Service provider does not provide this service, you can try either of these: www.sdforlaget.se/remoteip.asp or

http://www.changeip.com/ip.asp

Appendix E — Working with the 3050 ICP

The 3050 ICP adds features to your 5055 SIP phone that provide greater ease of use within a small office or retail establishment, and opportunities to improve how you interact with your customers.

Convenience

Mitel networks SIP phones connected to 3050 ICPs share a fast two, three or four-digit dialing for station-to-station calls.

Cost Savings

The 3050 ICP can help you to reduce or avoid many long-distance charges by using fixed-rate broadband IP to carry much of the voice traffic that would otherwise travel over the PSTN.

Customer Interaction

The 3050 ICP has an autoattendant that can be used to direct your callers to specific individuals or functions in your organization. For instance, if someone in your office usually handles inquires about customer orders, then your automatic attendant can be programmed to direct callers to this individual.

The 3050 ICP lets you assign voicemail accounts to each user for those occasions when they cannot answer the phone in person. Users record their personalized greeting using the 5055 SIP phone. When a caller leaves a message, it is delivered to the user's email inbox for later retrieval using their PC or a phone.

For further details on the Mitel Networks 3050 ICP contact your local Mitel representative or visit Mitel online at http://www.mitel.com

Appendix F — Frequently Asked Questions

How do I access the User Profiles?

User Profiles must be enabled on the phone, before they will work. Refer to User Profiles to learn how to do this.

Does my 5055 SIP Phone work behind a non-SIP compliant router?

It can - if you follow the steps outlined in Appendix D — Working with Firewalls

Where do I go to find latest versions of the 5055 firmware?

The latest version of the 5055 firmware is available on the TFTP server at: sipdnld.mitel.com. For detailed instructions on firmware upgrade, refer to

Upgrading the Firmware of the SIP Phone

Where can I find the latest 5055 SIP Phone documentation?

You can use a copy of Netscape Navigator or Internet Explorer to view and download an Adobe Acrobat compatible user guide by:

1. Point you browser to http://edocs.mitel.com.
2. Click the **User Guides** link to display the user guides page.
3. Locate the **Other** section – its has links for 5055 user and installation guides

Upon boot, the phone displays “PPPoE Initialize” and nothing else

The phone is configured to work with a DSL connection using PPPoE but this connection cannot be established.

1. Check the PPPoE login name and password in the Network Configuration Page
2. Make sure the DSL modem is plugged in, and powered up.
3. Verify that the SIP phone has got a valid IP address from the modem by pressing the **Menu** key, followed by the **Line 1** key.

The phone indicates a valid PPPoE connection by clearing the “PPPoE Initialize” message and proceeding with the rest of the boot-up process.

How do I find out the IP address of my 5055 SIP Phone?

1. Press the **Menu** key.
2. Press the **Line 1** key. The phone's IP and MAC address are displayed.

What version of boot and main firmware is currently installed on my phone?

1. Press the **Menu** key.
2. Press the **Line 2** key. The Main and Boot software versions are displayed.

What languages are currently available for my 5055 SIP Phone?

As of release 2.0:

- North American English
- North American French
- Latin American Spanish

You can select the display language using the User Configuration web page, or with the softkey menu system (Phone Settings).

Why does my phone show *NO REG*?

This message indicates that your phone has failed to register with a SIP registration server. Connection to a SIP registration server is necessary for your phone to be able to make and receive SIP calls.

For registration to be successful:

1. The SIP registration Server must be up and running.
2. You must have correctly entered the SIP Registration Server's URL (provided to you by your SIP service provider) in the **SIP Registry Server** field of the SIP Configuration Page.
3. If you SIP registration server is located on the Internet, then you must have a working connection to the Internet.

You must have correctly entered the SIP Authentication user name and password (provided to you by your SIP service provider) in the corresponding fields of the

4. User Configuration Page.

The time and date on my phone is not correct?

The phone can be configured to obtain its time automatically by consulting a Simple Network Time Protocol Server (SNTP), or it can be manually set to a specific time.

- If you are using SNTP to set the time automatically, then you must have entered the URL of a functioning SNTP server in the **Additional Servers** section of the Network Configuration Page. As many SNTP servers base their clocks on Coordinated Universal Time (Greenwich Mean Time), you may have to enter an offset value into the **Time Zone** field to accurately reflect your local time.
- If you set your phone's time manually, then you will have to reset it using the softkey menus (Phone settings), or the appropriate fields in the Feature Configuration Page.

Glossary

Term	Definition
3050	Mitel Networks(tm) 3050 Integrated Communications Platform (ICP)
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server
DSL	Digital Subscriber Loop
DTMF	Dual Tone Multiple Frequency
GMT	Greenwich Mean Time (the time at Meridian 0, which goes through Greenwich, UK)
HTTP	Hypertext Transfer Protocol
ICP	Integrated Communications Platform
ID	Identification
IP	Internet Protocol
LAN	Local Area Network
MAC	Media Access Control
NAT	Network Address Translation
PPPoE	Point-to-Point Protocol over Ethernet
PSTN	Public Switched Telephone Network
QOS	Quality of Service
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
SNTP	Simple Network Time Protocol
TCP	Transmission Control Protocol
TFTP	Trivial File Transfer Protocol
ToS	Type of Service
UDP	User Datagram Protocol
URL	Uniform Resource Locator
VLAN	Virtual LAN
WAN	Wide Area Network