1000i Series IP Phone (SIP)

User Guide

Please read this user guide carefully before operating your phone.
Retain it for future reference.

iPECS is an Ericsson-LG Brand

ERICSSON LG

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	Oct., 2020	Initial Release
1.1	Oct., 2020	Added the 1048ilss model
1.2	Sep., 2021	- Added the 1048idss model - Updated LCD configuration - Added Wi-Fi configurations (1040i/1050i only)
1.3	Oct.,2022	- Add the IPv6 information - Change the 1048ilss,1048idss connection quantity up to maximum two - Updated LCD configuration

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Regulatory Information

Important Safety Information

To prevent unexpected danger or damage please read this information before installing or attempting to repair you phone. Warning and Caution information is provided to alert the consumer of known dangers:

Warning: To reduce the possibility of electric shock, do not expose your phone to high humidity areas, such as a bathroom, swimming area, etc.

Warning

- Only trained and qualified service personnel should install, replace or service the phone.
- Do not spill liquid water on the phone. If so, disconnect the phone immediately and call the service center as this may result in fire or electric shock.
- If you see or smell smoke during use, disconnect the phone and call the service center immediately.
- If the power adaptor is used, do not touch the plug with wet hands. This may result in a fire, an electric shock or equipment damage.
- Do not use the phone during a thunderstorm. A lightning strike may result in fire, severe electrical or acoustic shock.
- Do not use the power adaptor if the power cord or wall outlet is damaged. This may result in fire or an electric shock.

Caution

- Ensure that children do not pull on phone cords. This may injure children or result in equipment damage.
- The earpiece houses a magnetic device that may attract pins or small metal objects. Keep the handset clear of such objects and check before use.
- Avoid placing the phone in an area that is excessively dusty, damp or subject to vibration.
- Choose a site for the phone that is well ventilated and dry.
- Do not plug multiple plug-packs into one power outlet. This may result in the plug overheating and may result in a fire or plug pack failure.
- Do not put heavy things on the phone.
- Do not drop or throw the phone.
- Static electricity discharge will damage electronic components.
- Keep out of direct sunlight and away from heat.
- No user-serviceable parts are inside. Do not insert a screwdriver or any metal objects into the phone. This may cause electric shock or damage the equipment and will render the warranty void.
- Clean the phone with a soft, dry cloth only. Do not use volatile liquids such as petrol, alcohol, or acetone as this may cause a fire or result in discoloration or damage to plastics. Do not clean with wax or silicon products as these may enter the equipment and cause operation to become unstable.

[EU] European Union Declarations of Conformity

Ericsson-LG Enterprise Co.,Ltd. declare that the equipment specified in this document bearing the " CE" mark conforms tothe Electromagnetic Compatibility Directive(EMCD,2014/30/EU) and Low Voltage Directive(LVD, 2014/35/EU).

Copies of these Declarations of Conformity (DoCs) can be obtained by contacting your local sales representative.

[USA/CSA] FCC/IC Interference Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules.

These limits are designed to provide reasonable protection against harmful interference in a residential installation.

This device complies with part 15 /RSS-GEN of the FCC/IC rules. Operation is subject to the following two conditions:

(1)This device may not cause harmful interference; and (2)This device must accept any interference received, including interference that may cause undesired operation.

(1)l'appareil ne doit pas produire de brouillage, et (2)l'utilisateur de l'appareil doit accepter tout brouillage radioélectrique subi, même si le brouillage est susceptible d'en compromettre le fonctionnement.

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation.

If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This Class B digital apparatus meets the requirements of the Canadian Interference-Causing Equipment Regulations, CAN ICES-3(B)/NMB-3(B)

Cet appareil numérique de la classe B respecte toutes les exigences du Règlement sur le matériel brouilleur du Canada.

CAUTION : Any changes or modifications in construction of this device which are not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

ATTENTION: Tout changement ou modification dans la construction de cet appareil qui ne sont pas expressément approuvé par la partie responsable de la conformité pourraient annuler l'autorité de l'utilisateur à utiliser l'équipement.This Class B digital apparatus complies with Canadian ICES-003.

Disposal of your old appliance

- When the crossed-out wheeled bin symbol is attached to a product, it means the product is covered by the European Directive 2012/19/EU.
- All electrical and electronic products should be disposed of separately from the municipal waste stream via designated collection facilities appointed by government or the local authorities.
- The correct disposal of your old appliance will help prevent potential negative consequences for the environment and human health.
- For more detailed information about disposal of your old appliance, please contact your city office waste disposal service or the shop where you purchased the product.

Before Use

About this Guide

This guide is intended for users of the 1010i, 1020i, 1030i, 1040i and 1050i iPECS IP Phone. The guide provides user-related information.

Introduction of the Document

This guide explains how to install and correctly use 1010i, 1020i, 1030i, 1040i and 1050i phones. Product figures and screen images used in this guide may be changed without notice for quality improvement.

Explanation of Symbol or Word

The following are symbols used in this guide. Information presented with these symbols must be observed for the correct use of your IP Phone.

CAUTION

This mark warns of a situation in which light injury or product damage (hardware, software or data damage) may occur if the instructions are not followed.

NOTE

A note provides additional explanation, emphasis on important information or reference for related information, which is required for the correct use of the phone.

1 Introduction

1.1 General

Your telephone is connected to an advanced-technology, highly versatile, fully featured telecom system designed to make office communications easy and productive.

Employing state-of-the-art Voice over Internet Protocol (VoIP) technology, voice and data converge on a single IP packet network. Because each iPECS Phone is an IP appliance, it can be moved easily throughout the office wherever there is a LAN connection and maintain normal operation.

1.2 Feature Information

We have taken every effort to make this user guide simple and straightforward. The guide starts with the simpler operations of the iPECS Phone Speakerphone, moves on to receiving and placing calls, and then to more advanced features. Each section includes a brief and basic description of each feature and step-by-step operation instructions.

The operations shown in this guide use the System's base default Numbering Plan. Your specific Numbering Plan may be different. Some features may not be available for you to access or may be subject to certain limitations based on the set-up of the System. Please consult with your System Administrator for further information.

NOTE
Under certain operating conditions, this equipment may not be able to make emergency calls. Alternative arrangements should be made for access to emergency services.

1.3 Feature Groupings

Every effort has been taken to divide the features into a logical and consistent sequence so that features can be quickly located. Features have been divided into groups, Receiving Calls, Placing Calls, etc. Within each group, features are arranged according to the difficulty of operation and frequency of use.

1.4 iPECS 1000i Phone Description

The iPECS 1000i series phones incorporate the latest in VoIP technology and user interface to provide a cost effective, simple to use, productive communications tool.

Each 1000i phone includes an LCD, fixed and flexible buttons with LEDs, Navigation keys, and full-duplex speakerphone. As shown in the layouts that follow the size of the LCD, number of flexible buttons and softkeys varies for each model. Refer to Appendix A: 1000i Series Model Specification for details.

- The 1010i includes a 4-line 132 by 64 dot graphic LCD, 4 flexible buttons and 3 softkeys.
- The 1020i includes a 4-line 132 by 64 dot graphic LCD, 8 flexible buttons and 3 softkeys.
- The 1030i includes a 5-line 320 by 240 dot graphic LCD, 6 flexible buttons and 3 softkeys.
- The 1040i includes a 6-line 480 by 320 dot graphic LCD, 8 flexible buttons and 3 softkeys.
- The 1050i includes an 8-line 480 by 272 dot graphic LCD, 12 flexible buttons and 3 softkeys.

NOTE

Features and functions of your iPECS are accessed using the Fixed or Flexible buttons, Soft keys or dial codes entered from the dial pad.

1.4.1 Phone Layout and Functions

1.4.1.1 Phone Layout

The figures below present line drawings of the front of each 1000i series phone identifying major elements of the user interface. As shown, the layouts of the 1010i, 1020i, 1030i, 1040i and 1050i are slightly different.

1010i Layout

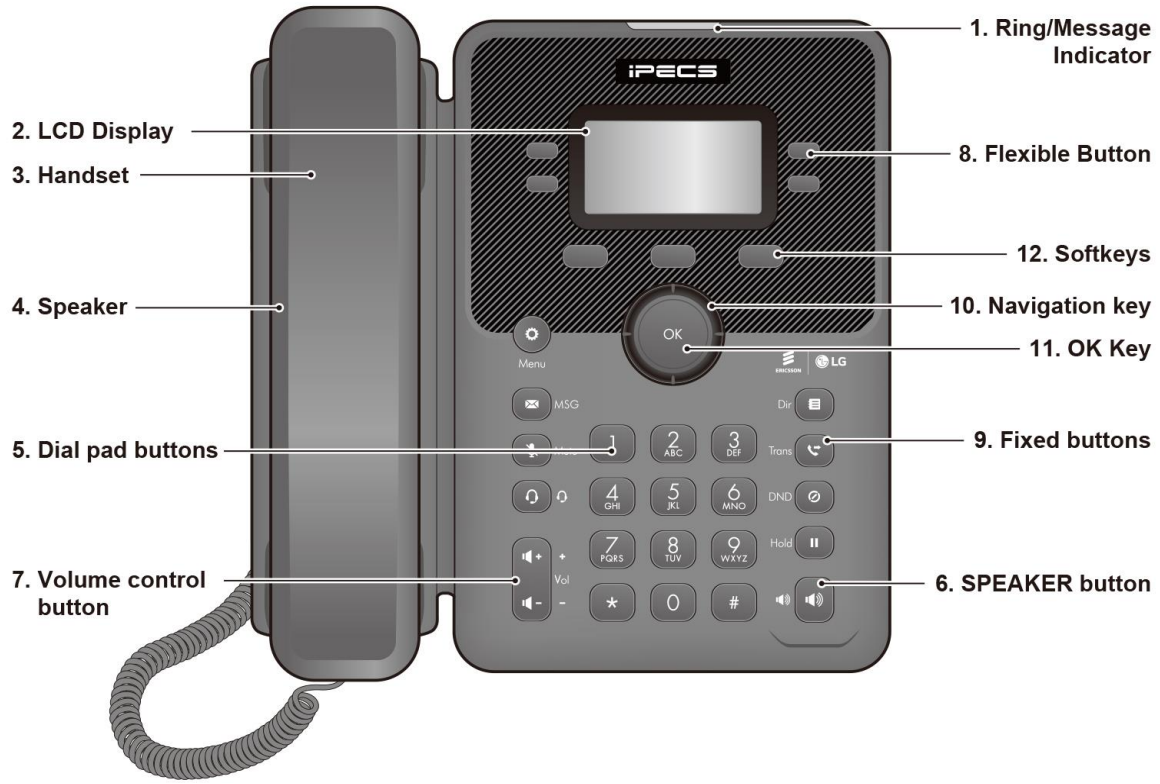


Figure 1.4.1-1: 1010i Front

1020i Layout

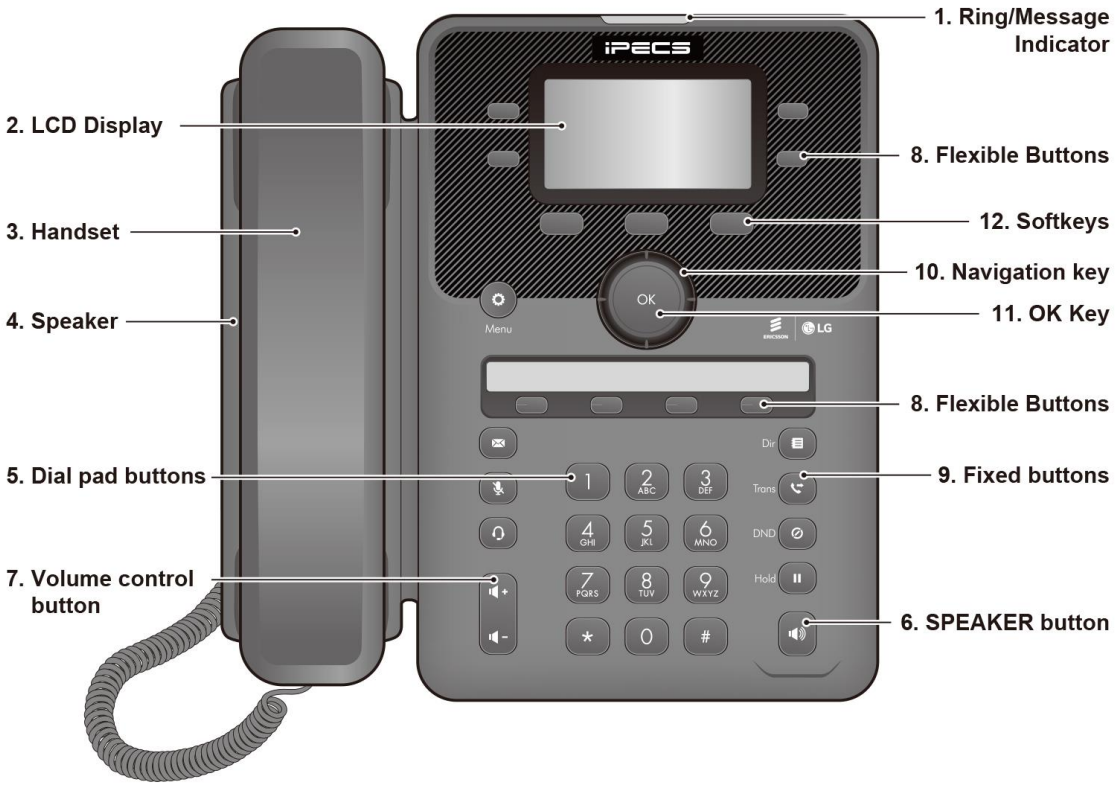


Figure 1.4.1-2: 1020i Front

1030i Layout

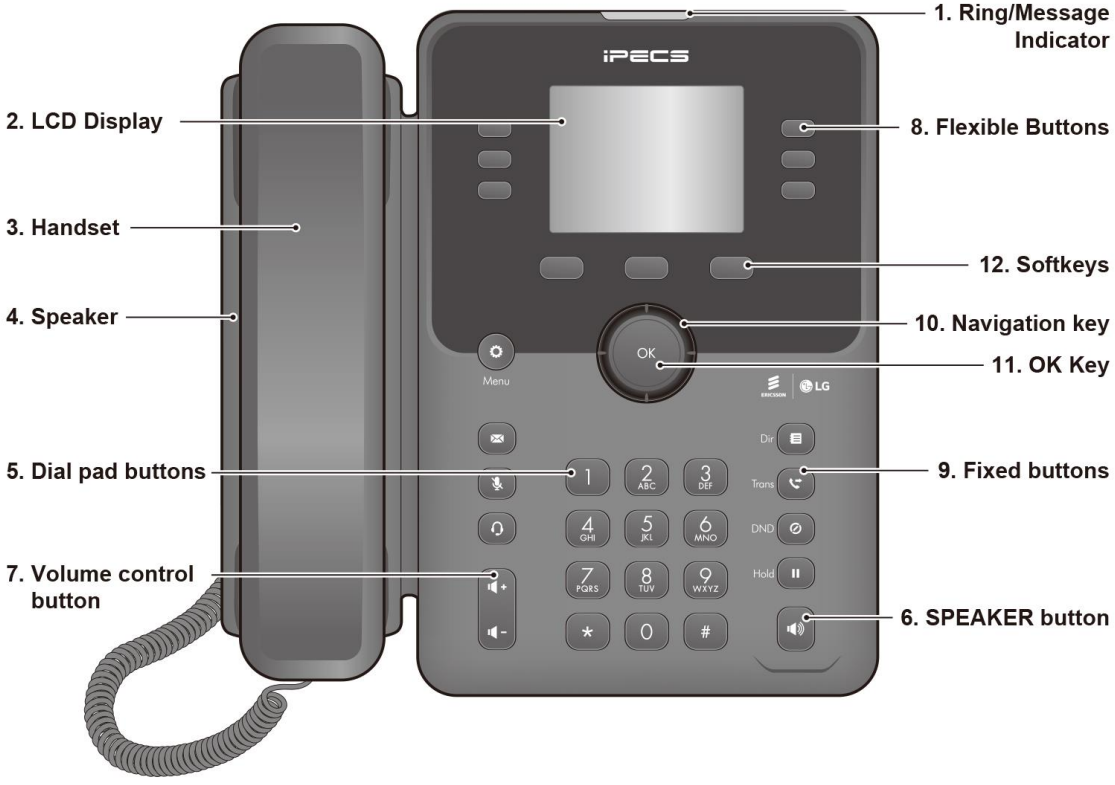


Figure 1.4.1-3: 1030i Front

1040i Layout

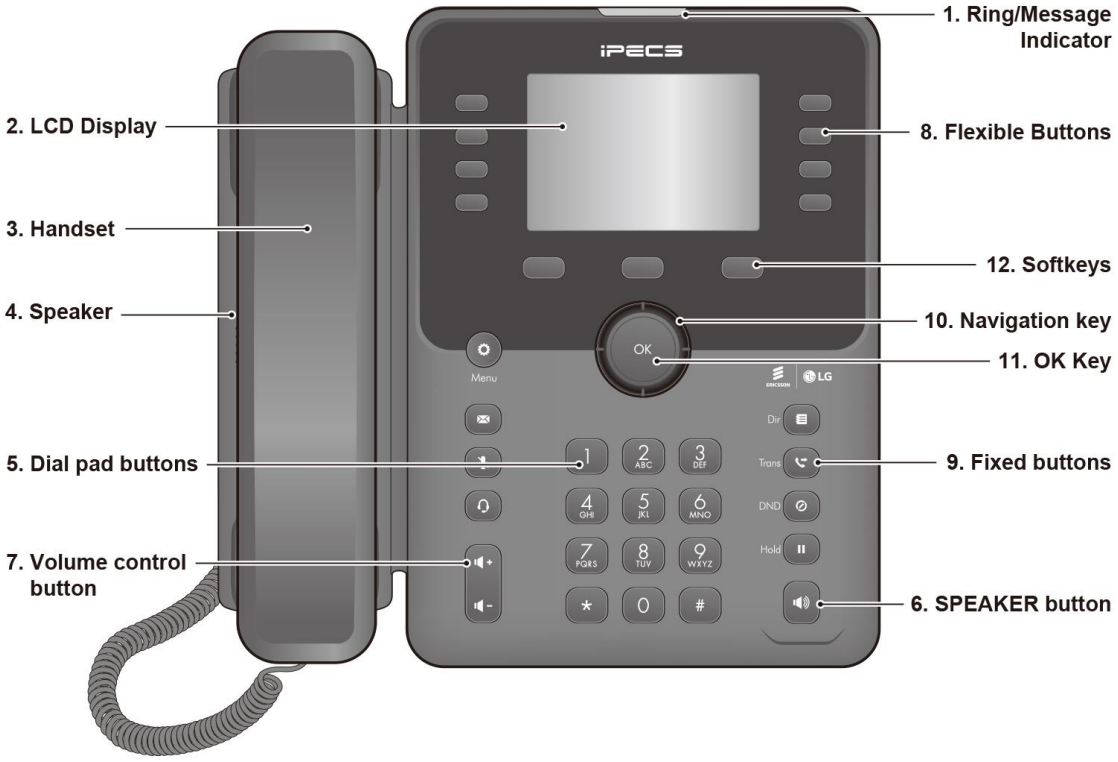


Figure 1.4.1-4: 1040i Front

1050i Layout

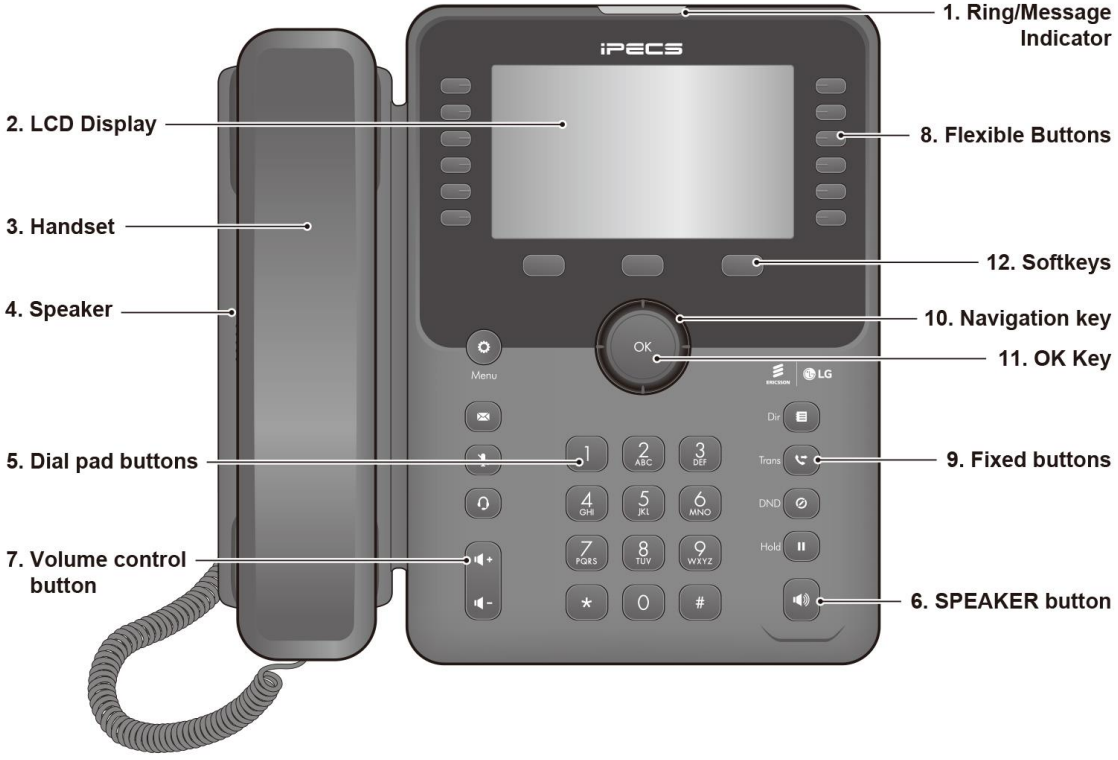


Figure 1.4.1-5: 1050i Front

1.4.1.2 Functions

Below is a brief description of the elements called in the 1000i series phone diagram above.

1. **Ring/Message Indicator:** Flashes when receiving a new call or you have a new message waiting.
2. **LCD Display:** Displays your phones status, dialing directories, and text message information.
3. **Handset:** Use to answer a call using the handset.
4. **Speaker:** Delivers ringing and other signals, and lets you hear the caller's voice in Speakerphone mode.
5. **Dial pad buttons:** Use to dial a number, select a menu item, or input a value.
6. **Speakerphone button:** Toggles the IP Phone Speakerphone on and off. The button illuminates when the Speakerphone is active, or the phone is in menu mode. (Except 1010i).
7. **Volume control button:** Use to adjust Ring, Headset, Handset, and Speaker volume.
8. **Flexible buttons:** Assigned as Line or feature access button.

- 1010i has 4-flexible buttons.
- 1020i has 4-flexible buttons with 3 pages and 4 flexible buttons with a paper label, total 16 buttons.
- 1030i has 6-flexible buttons with 3 pages, total 18 buttons.
- 1040i has 8-flexible buttons with 3 pages, total 24 buttons.
- 1050i has 12-flexible buttons with 3 pages, total 36 buttons.

9. Fixed buttons

- Menu ( ) button: Accesses the menu for display and changes to the IP Phone configuration.
- Transfer ( ) button: Transfers the current active call. The button switches the dial pad input mode (A, a, * or 1). The dial pad mode displays in the upper-right corner.
- Directory ( ) button: Accesses your private or public directory for speed dialing.
- Message ( ) button: When the MWI LED indicates you have a message, use to access your Voice Mailbox.
- DND ( ) button: Use to activate DND (Do Not Disturb) so that your phone will not ring. The button also deletes the last character of an input in the menu mode.
- Headset ( ) button: When using a headset, this button toggles the headset state. When the headset is active, the button LED illuminates red and the Headset Icon will be displayed for 1010i.
- Hold ( ) button: Use to place a call on Hold. Also, use to access a held call. In menu mode, it saves your inputs.
- MUTE ( ) button: Toggles audio from the microphone to the connected party on and off. While the IP Phone is idle, used as a redial function.

10. Navigation key

- Left: Use to move the previous group of softkey.
- Right: Use to move the next group of softkey.
- Up: Scrolls through the phone's menu options upward.
- Down: Scrolls through the phone's menu options downward.

11. **OK Key:** Selects the highlighted choice when navigating through a menu. In 1030i, 1040i and 1050i, during a call, Pop-up display and call information can be toggled using "OK" button. Pop-up disappear and call information display in the top bar.
12. **Softkeys**: Softkeys are interactive, changing function based on the status of the phone.

1.5 1000i Installation

The 1000i Phones are shipped with the 1000i Phone, handset and coil cord, adjustable foot-stand, 1.5-meter (~5-foot) Cat 5 cable terminated in RJ-45 connectors, and the Quick Guide.

An optional AC/DC adaptor is available for use when the network connection does not provide power to the phone (support IEEE 802.3af PoE -Power over Ethernet-).

Several other options including DSS(Refer to 1.6 DSS Overview & Installation)and wall mounting(Refer to 1.7 Wall Mount Installation) are available.

The phone has a number of connections available including the handset and LAN connection, which are required. Connections to your desktop PC, headset and optional power adaptor are incorporated in the phone design.

To install the phone;

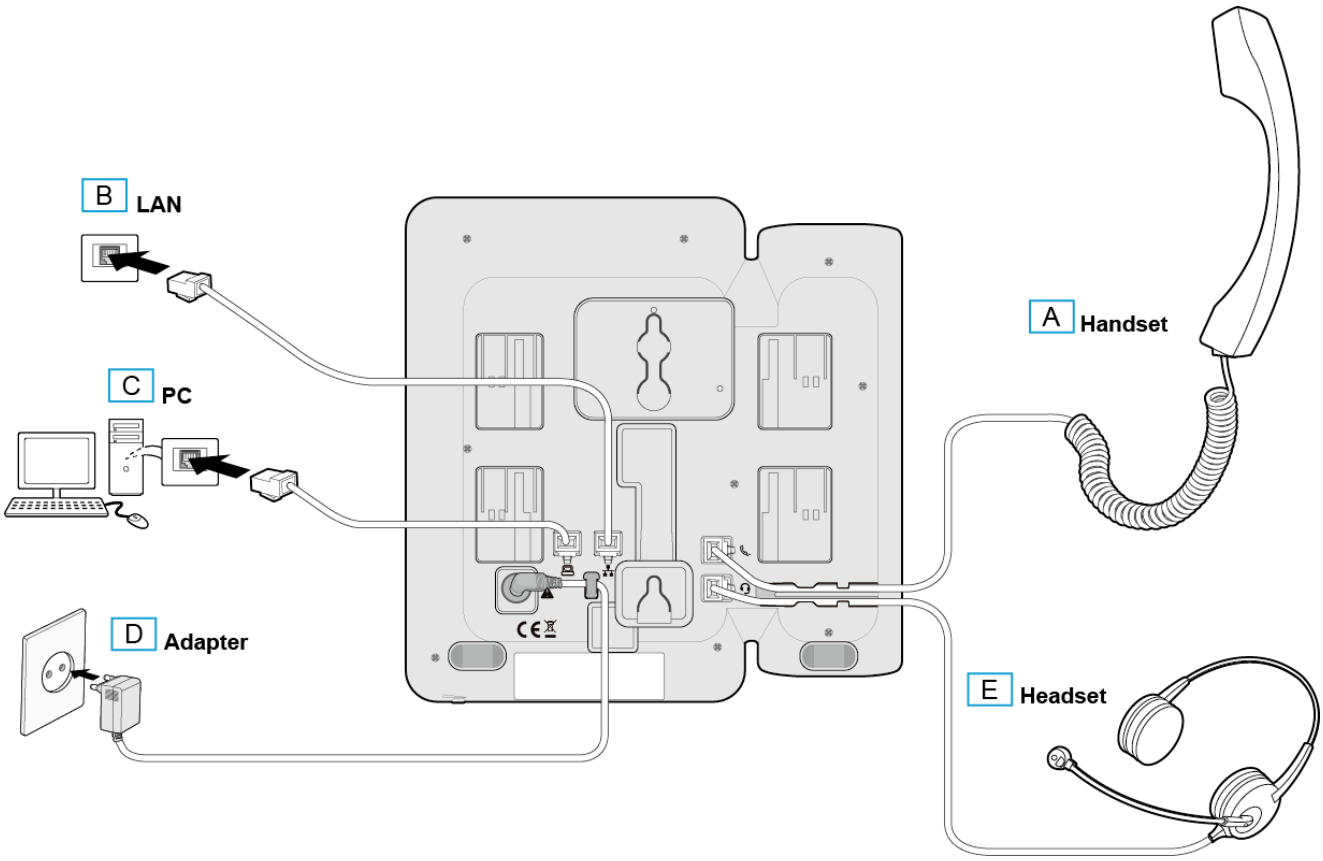


Figure 1.5: 1000i Installation

1) If needed install the DSS assembly as described in “1.6 DSS Overview & Installation” section.
2) Insert the connectors of the coil cord into the handset and phone handset port as shown in A.
3) Insert the provided Cat 5 cable into the network connection of the phone and the LAN wall jack as shown in B.
4) When needed, your desktop PC can be connected to the PC LAN port of the phone. Connect a Cat 5 cable (not provided) between the PC’s LAN port and the PC LAN port of the 1000i phone as shown in C.

5) If your LAN port does not support Class 2 Power over Ethernet (PoE), you must install the optional AC Power adaptor. Note do not connect the power adaptor if your port does support PoE. Connect the DC output of the adaptor into the phone and the AC plug into a standard AC power outlet as shown in D. Consult with your system administrator to determine if you must install the AC power adaptor. You may need to contact your local ELG-Enterprise representative for the appropriate power adaptor for your region.

Caution

- To power your phone, use either PoE or the AC adaptor, not both.
- Use the Ericsson-LG Enterprise approved AC power adaptor only. The AC adaptor is ordered separately.

6) A compatible headset may be connected to the Headset jack in the bottom of the phone. As shown in E, connect the headset plug into the headset jack. The 1000i phones are compatible with several headsets. Please consult with your local Ericsson LG-Enterprise representative for a list of headsets compatible with the 1000i.

The 1040i/1050i supports USB Headset. If a USB Headset is plugged in, the Headset function operates as a USB Headset.

NOTE: If you plug in or unplug the USB Headset during a call, it will operate from the next call.

7) The phone includes a foot stand that can be installed to provide a 35° or 50° angle of the face of the phone. If the phone is not wall mounted, install the foot stand at the desired angle. If wall mounting, do not install the foot stand.

NOTE

Once installation is complete, the phone may require configuration of the network parameters before becoming operational. These parameters should be modified by a trained technician only.

1.6 DSS Overview & Installation

1.6.1 1024idss overview

The 1024idss provides the 24 additional flexible buttons. The flexible buttons on the DSS are used as with the flexible buttons on the phone proper and can be assigned for any of the flexible button functions. One 1024idss can attach to an 1020i, 1030i ,1040i or 1050i phone, and can be placed on your desktop or can be wall-mounted with the phone.

The 1024idss is described as follows and shown in below figure.

- The 1024idss has 24 flexible buttons with tri-color LED. The 1024idss uses a paper label for the button designations.

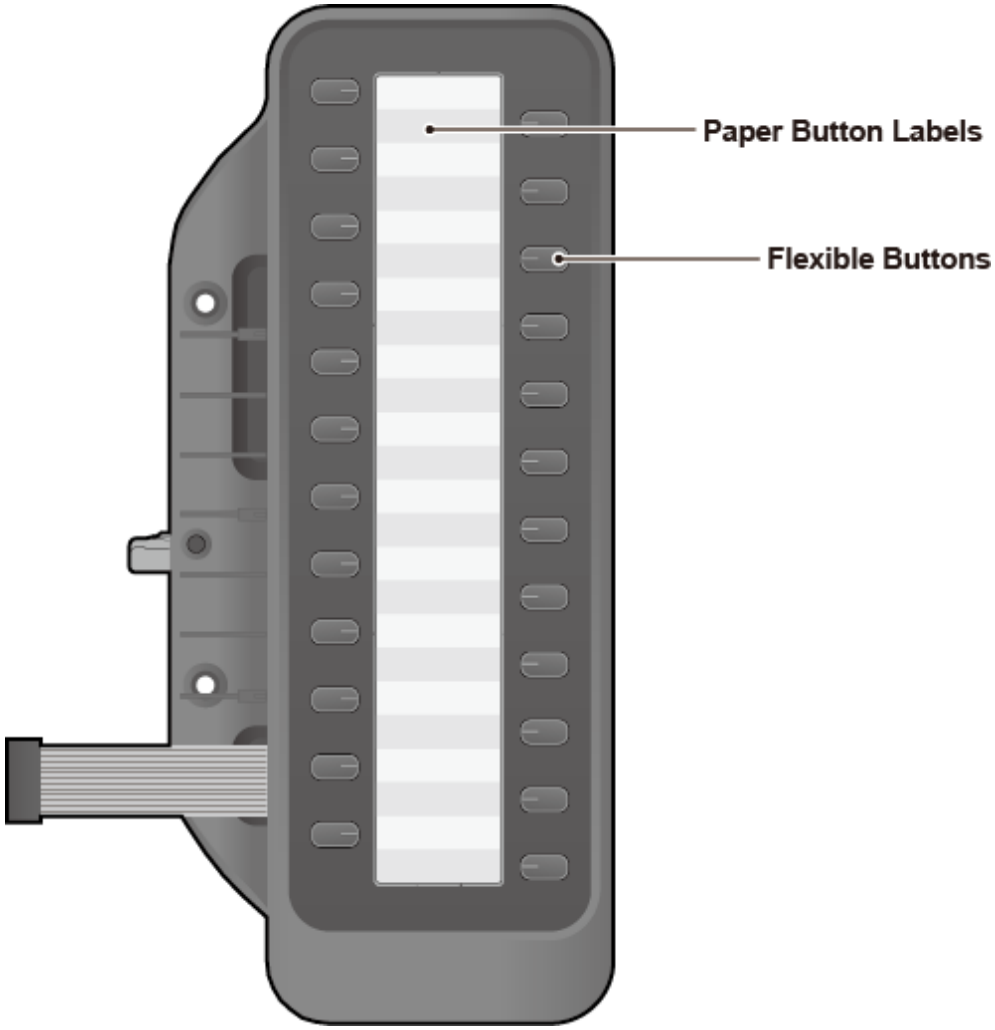


Figure 1.6-1: 1024idss

Caution

To avoid damaging the Phone or DSS, remove power from the phone before installing a DSS.

1.6.2 1048ilss overview

The 1048ilss provides the 2x24 flexible buttons that can be programmed to connect with an IP phone. You can connect up to two 1048ilss to the phone. (except the 1010i), and can be placed on your desktop or can be wall-mounted with the phone.

NOTE

The iPKTS phone can connect up to two 1048ilss, but the 1048ilss cannot be used with 1024idss or 1048idss.

The 1048ilss is described as follows and shown in below figure.

- The 1048ilss provides the 2x24 flexible buttons with tri-color LED

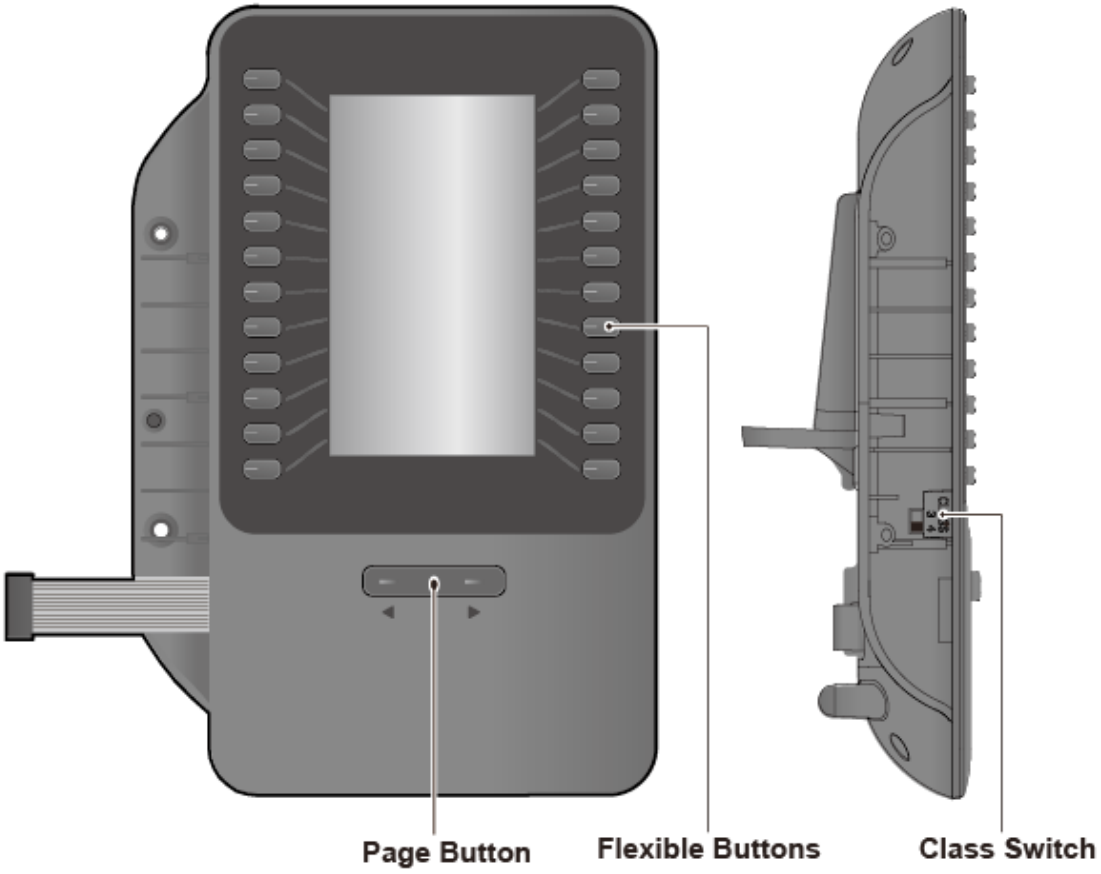


Figure 1.6-2: 1048ilss

Caution

Disconnect the power cable or remove the network cable to ensure that your phone is Off. Only a peripheral device can be connected to the AEM port. (1048ilss or EHSA)

USB Charging from the Phone

When charging using the phone's USB port, be sure to connect the Adaptor to the phone.(1040i,1050i only)

1.6.3 1048idss overview

The 1048idss provides 48 flexible buttons that can be programmed to connect with an IP phone. You can connect up to two 1048idss to the phone(except the 1010i), and can be placed on your desktop or wall-mounted with the phone.

NOTE

The iPKTS phone can connect up to two 1048idss, but the 1048idss cannot use with 1024idss or 1048ilss.

The 1048idss is described as follows and shown in the below figure.

- 1048idss: 48 flexible buttons with tri-color LED

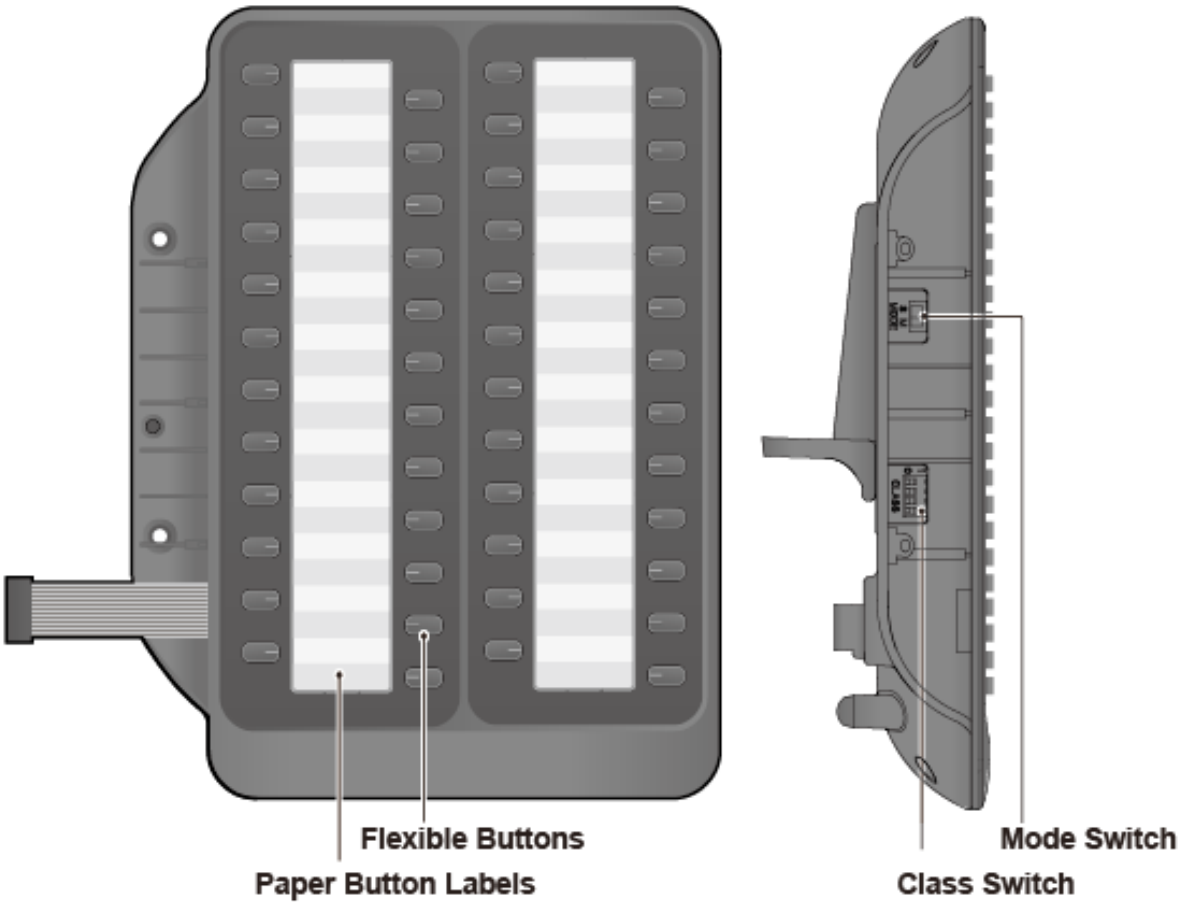


Figure 1.6-3 1048idss

Caution

Disconnect the power cable or remove the network cable to ensure that your phone is Off. Only a peripheral device can connect to the AEM port. (1048idss or EHSA)

USB Charging from the Phone

When charging using the phone's USB port, be sure to connect the Adaptor to the phone. (1040i,1050i only)

1.6.4 DSS Installation

The DSS should be installed on the phone prior to making any connections to the phone. This simplifies the installation for 1030i Phone and 1024idss. You can install the 1048ilss,1048idss in the same way.

For more detailed installation instructions, Scan the QR code that is printed on the product label of 1024idss or 1048ilss,1048idss to refer to a Quick installation Guide.

To install the DSS

1) First, assure the phone is NOT connected to power. For convenience, you may wish to remove the handset and foot-stand; this simplifies the DSS installation.
2) Align the DSS and phone as shown in the drawing.
3) Connect the DSS Connecting Cable to the phone.
4) Push the DSS into the phone until the DSS is fully engaged.
5) Insert the screws provided through the screw tabs in the DSS and tighten the screws into the phone. Do not over-tighten.

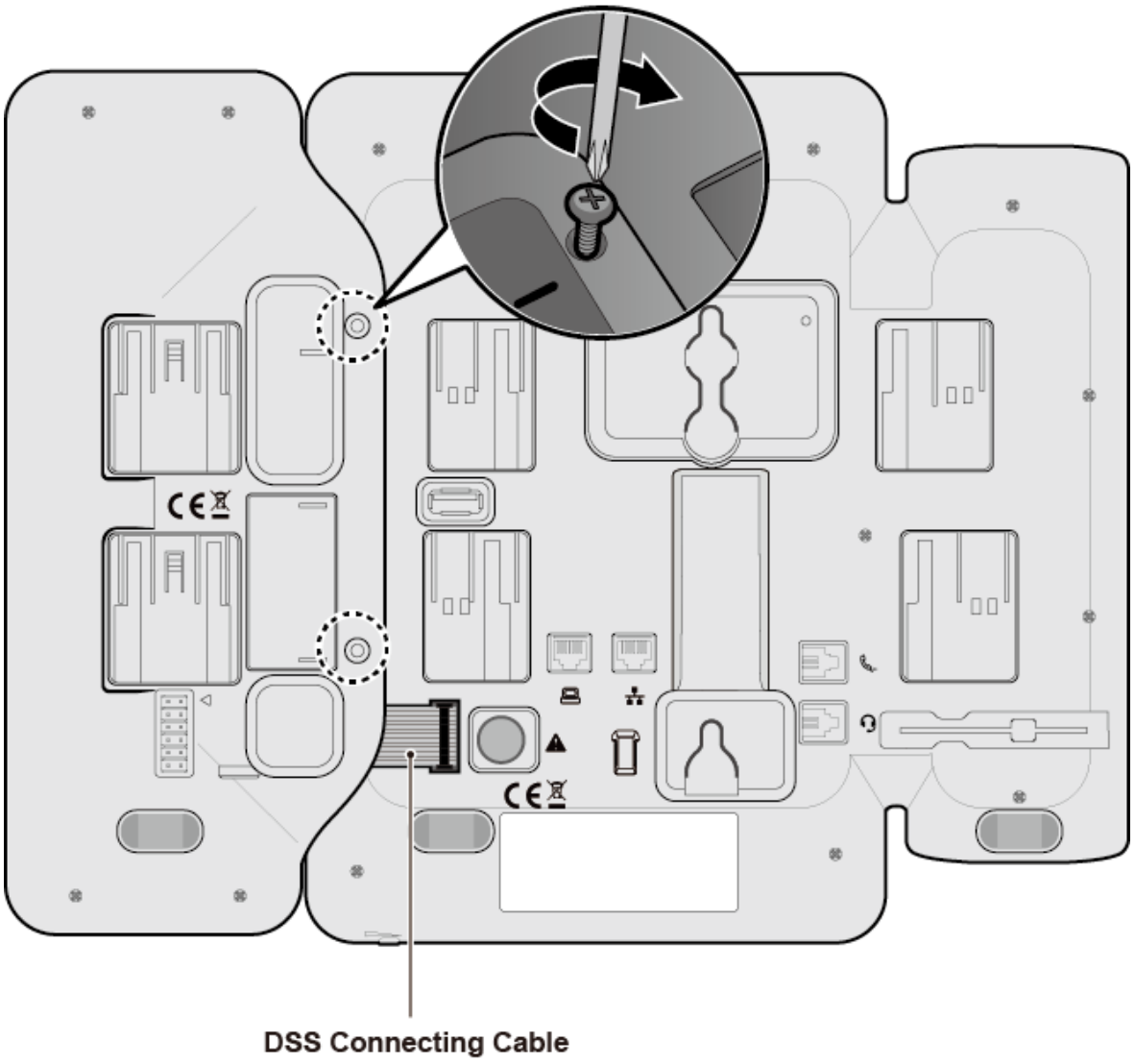


Figure 1.6-4: 1024idss Installation

1.7 Wall Mount Installation

The following instructions describe the wall mount installation. Note the foot-stand must not be installed.

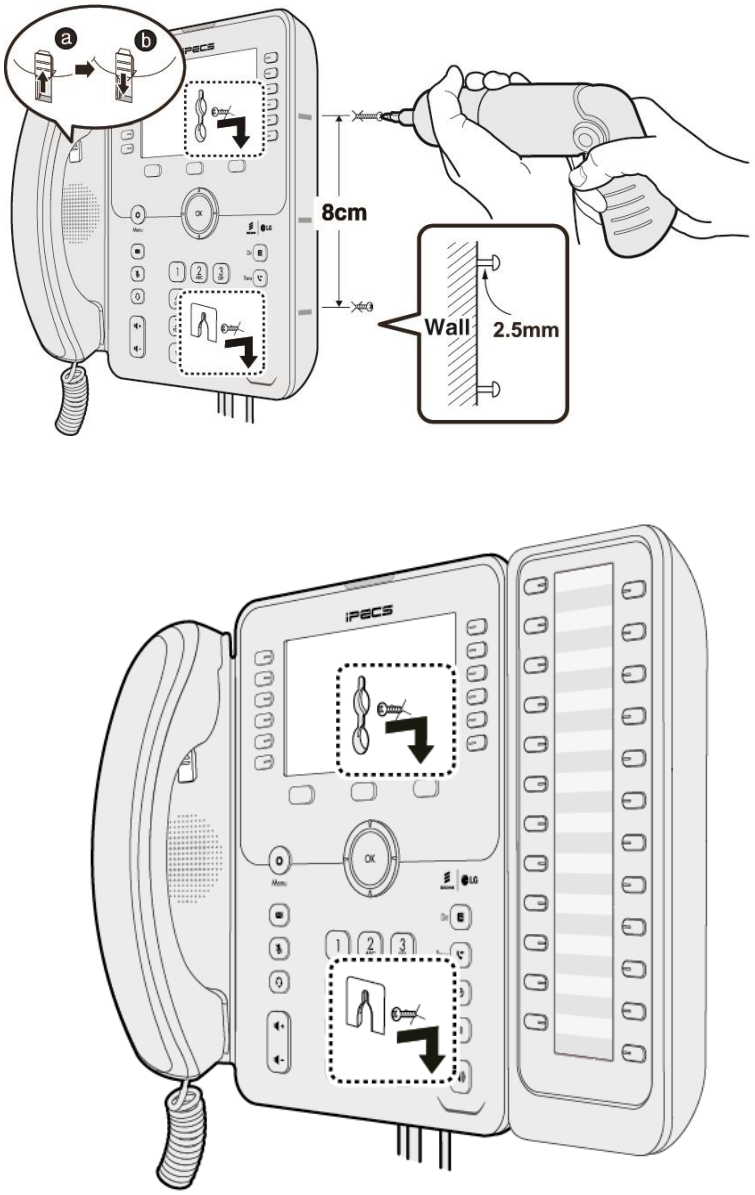


Figure 1.7: Wall Mount Installation

1) If installed, remove the foot-stand from the Phone and DSS.
2) Connect all wiring to the Phone as described in "1.5 1000i Installation" section. Connect the Phone side wiring only.
3) If required, install DSS as described in "1.6 DSS Overview & Installation" section.
4) Mark and drill two 7mm holes for plastic wall anchors (not included).
5) Insert the two anchors into the holes and insert and tighten the two screws (not included) leaving about 2.5 mm (1/8-inch) exposed.
6) Align the key holes in the phone over the screws and slide the phone down assuring the phone is secure.

7) Remove, reverse and re-install the handset hook so that the hook catches the groove in the handset receiver.
8) Hang the handset up on the handset hook.
9) Complete all wiring connections as described in "1.5 1000i Installation" section.

NOTE

It may be necessary to remove the phone and tighten or loosen the screws for a secure mounting.

1.8 EHS Adaptor

Electronic Hook Switch (EHS) Adaptor is a device which electronically connects a EHS Base Set and Ericsson-LG Enterprise phone. This EHS Adaptor enables remote control with the compatible EHS Base Set and the 1000i Series (except 1010i).

Compatible EHS Base Set Model List

Below are the recommended EHS Base Set models for your phone. For further information on installation see the EHS Adaptor Quick Installation Guide which is packed with it.

- **Jabra (IQ Protocol):** PRO P20, PRO 925, PRO 9450
- **Plantronics (PSB Protocol):** Savi W720, Savi W740-M, CS540
- **Sennheiser** (DHSG Protocol): D10, DW Pro

NOTE

Ericsson-LG Enterprise cannot guarantee other model except the above list and this list will be updated if we have finished testing the new one anytime.

1.9 Wi-Fi dongle

The 1040i and 1050i have a USB port so you can connect to a wireless network using a Wi-Fi dongle and provides LCD or Web menu for Wi-Fi setting. The Wi-Fi setting menu is only displayed when the Wi-Fi dongle is installed, so you must first plug the dongle into the USB port on the back of the phone. In the LCD menu, setting is possible even if there is only an AP that can be connected.

All network and server settings except "Wi-Fi Settings" related to Wi-Fi connection use the wired network settings together.

Supported models and version

- 1040i and 1050i (Version R1.1.0 or later)

Compatible Wi-Fi dongle Lists

- **TP-LINK Archer T3U** (AC1300 Mini Wireless MU-MIMO USB Adapter, Ver:1.0)
- **TP-LINK Archer T4U** (AC1300 Wireless Dual Band USB Adapter, Ver:3.0)
- **D-Link DWA-182** (Wireless AC1300 MU-MIMO Wi-Fi USB Adapter, Revision D)

NOTE

Ericsson-LG Enterprise cannot guarantee other model except the above list and this list will be updated if we have finished testing the new one anytime.

1.10 Entering Text

1.10.1 Phone Entering Text

Several features available to your phone require text entry. Station and Speed dial name assignment, and Intercom Text Messaging all require character entry.

Characters are entered by pressing the dial pad digit with the character multiple times based on the character position on the button.

For example;

The letter ‘C’ is the third letter on the digit ‘2’ thus, to enter a ‘C’, the dial pad digit ‘2’ would be pressed 3 times.

If you want to enter the ‘C’ consecutively, wait a second to be moved the cursor to the right side of ‘C’ or press the right key of the navigation if it has navigation keys.

You can enter upper or lower case letters, symbols, or numeric characters. The entry mode is controlled with the Transfer(PGM) button.

The entry mode is indicated in the LCD of your phone as A, a, * or 1.

- The DND button deletes the last character. Or you may use the DELETE Softkey.
- The TRANSFER (PGM) button changes the input mode (A, a, * or 1) or you may use the MODE Softkey to change modes.

The chart below maps each dial pad digit to the corresponding character, symbol, and numeric entry.

NOTE

1000i series phone's Character Entry Chart (D1: 1st Digit, D2: 2nd Digit, D3: 3rd Digit, D4: 4th Digit)

Table 1.10.1: Phone Entering Text

Dial Pad Digit	Entry Mode				
	Upper Case A	Lower Case a	Symbols ✳	Numeric 1	IP address
	Button Depressions				
	D1, D2, D3, D4	D1, D2, D3, D4	D1, D2, D3, D4	D1	D1, D2, D3, D4
1	@ : /	@ : /	@ : /	1	1
2	A B C	a b c	| $ %	2	2 A B C
3	D E F	d e f	^ & _	3	3 D E F
4	G H I	g h i	( ) ?	4	4
5	J K L	j k l	- + /	5	5
6	M N O	M n o	< > =	6	6
7	P Q R S	p q r s	; :	7	7
8	T U V	t u v	’ ” `	8	8
9	W X Y Z	w x y z	{ }	9	9
0	. , ? !	. , ? !	. , ? !	0	0
*****	. *	. *	. *	*	. (IPv4)
#	Space (‘ ‘)	Space (‘ ‘)	# [ ]		: (IPv6)

2 Using the LCD & Speakerphone

2.1 Using the LCD

Each 1000i model has a different display. The top line of each display shows various icons as shown in the Icons chart below.

The 1010i and 1020i have a 4-line display. The upper section, which consists of a single line, displays station number, various icons and time.

The 1020i has flexible button page number. The middle section displays the label for each flexible button. The fourth line displays the interactive softkey menu to guide you through feature access and User Program selections. The display indicates the function for each softkey based on the status of your phone. In some cases, there may be more than three available softkey functions. In this case, a right arrow displays on the right side of the menu line. The right Navigation key scrolls to the next set of three functions, and the left key moves to the previous menu choices.

The 1030i, 1040i and 1050i have a 5-line, 6-line and 8-line Liquid Crystal Display (LCD), respectively. The upper section, which consists of a single line, displays station number, various icons, time and flexible button page number.

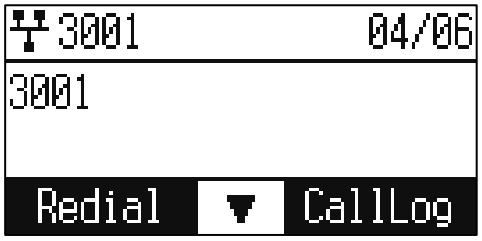
The middle section displays the label for each flexible button or a monthly calendar displays by the OK key. This section of the display also provides the called or calling name or number display, feature status, etc. The lower section, which consists of a single line, is the interactive menu of three softkeys described in the previous paragraph.

The 1030i, 1040i and 1050i has a screen saver function. The screen saver has two kind mode, Clock or Off. If not used, the screen saver will be activated after 10 minutes. This value or mode can be changed on the web manager.

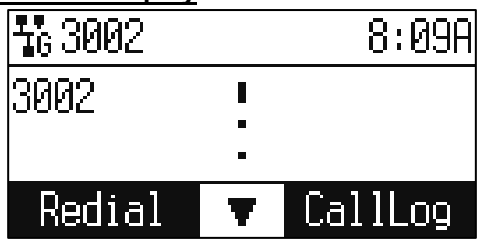
2.1.1 1000i Series Phone Displays

Below is a figure of Display layout of each 1000i phone.

1010i Phone Display

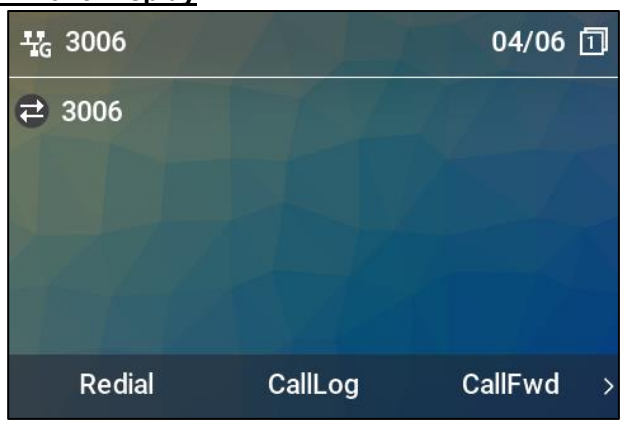


1020i Phone Display



1030i Phone Display

1040i Phone Display



1050i Phone Display

2.1.2 Upper line and Flexible button icons

Below is a chart list of each icons and their meaning.

2.1.2.1 Upper line Icons

Table 2.1.2-1: 1010i, 1020i Icons Description

Icon	Description
	The MUTE icon indicates the microphone is On and Off. (1010i only)
	The Headset icon indicates headset state, when shown the headset is active. (1010i only)
	Speakerphone icon, indicates Speakerphone activation state
	Hold icon, indicates hold state.
	LAN icon indicates the status of the LAN connection to the IP Phone. If the network speed is connected with Gigabit, the Giga-Icon () will be displayed. (1020i only)
	Missed call icon, indicates one or more missed calls.
	Call icon indicates call information during a call.
	Forward icon, indicates calls to your station are forwarded.
	The SRTP icon indicates SRTP call.
	Voice Message icon, indicates one or more new voice messages.
	Text Message icon, indicates one or more new text messages.
	EHS icon is displayed when EHSA is installed. If there is an error between EHSA and EHS headset, it will be changed to the EHS OFF icon () (1020i only).
	This icon is used to move another softkey menu

	This icon is used to select or save the current menu.

Table 2.1.2-2: 1030i, 1040i, 1050i Icons Description

Icon	Description
	LAN icon indicates the status of the LAN connection to the IP Phone. If the network speed is connected with Gigabit, the Giga-Icon () will be displayed.
	Missed call icon, indicates one or more missed calls.
	Call icon indicates call information during a call.
	Forward icon, indicates calls to your station are forwarded.
	The SRTP icon indicates SRTP call.
	Voice Message icon, indicates one or more new voice messages.
	Text Message icon, indicates one or more new text messages.
	EHS icon is displayed when EHSA is installed. If there is an error between EHSA and EHS headset, it will be changed to the EHS OFF icon ().

Table 2.1.2-3: 1040i, 1050i Wi-Fi Icons Description

Icon	Description
	The wireless connection status and signal strength are displayed on the top bar of the screen. (Very Poor[] / Poor / Fair / Good / Excellent[])
	Indicates that the wireless is disconnected when using a wireless connection.
	It is displayed when the Secured AP is displayed in the list.
	Appears when already Saved APs are displayed in the list.

	Appears when the currently connected AP is displayed in the list.
	Indicates the Signal Strength of APs shown in the list.
	Indicates the wireless band of the APs shown in the list. ( : 2.4G only,  :5G only,  : 2.4G/5G Both)

2.1.2.2 Flexible button icons (1030i, 1040i, 1050i)

Table 2.1.2-4: 1030i, 1040i, 1050i Flexible button icons Description

Icon	Description
	Not Assigned
	Available (line) icon indicates the line is available for use.
	Unavailable (line) icon indicates the line is not available for use.
	Ring (line) icon indicates the line has an incoming call.
	Busy (line) icon indicates the line has an active call.
	DSS/BLF (line) icon indicates the line has DSS/BLF function
	Conference (line) icon indicates the line is presently in a conference call.
	Hold (line) icon indicates there are calls on hold for the line.
	DND (line) icon indicates the line is in DND and will not notify you of incoming calls on the line.
	Forward (line) icon indicates calls to the line are forwarded to another destination.

	Message (line) icon indicates there are messages waiting for the line.
	Speed icon indicates the speed number is assigned in the line.
	Phonebook icon indicates the Phonebook is being displayed.
	Feature or function icon indicates the feature or function is assigned in the line.

2.2 Using the Speakerphone

Activate the iPECS Speakerphone at any time (except for Paging) in place of lifting the handset to receive or place calls.

Additionally, you may switch between the handset and Speakerphone during a call. By keeping the handset Off-hook, the Group Listen feature is activated, which provides incoming audio over the speaker with outgoing audio from the handset. This permits a local 'group' to listen to both sides of the conversation without interfering with the conversation.

While in a conversation using the Speakerphone, Mute the microphone using the MUTE button. When Mute is active, the MUTE button LED will be illuminated. To turn off Mute, press the MUTE button again.

With Auto Speaker Select, activating or accessing a feature, or placing a call by pressing a flexible button will activate the Speakerphone automatically.

To use the Speakerphone to answer or place a call

- ✓ Press the SPEAKER button, the phone goes off-hook and sends audio to the speaker and from the microphone.

To use the Headset to answer or place a call

- ✓ Press the HEADSET button, the phone goes off-hook and sends audio to/from the headset.

To control the volume

- ✓ During a call, press the VOLUME + or – button.

To mute the microphone

- ✓ During a call, press the MUTE button.

To turn off Mute (turn the microphone ON)

- ✓ Press the MUTE button. Mute LED will be turn off.

To activate Group Listen while on an active handset call

1) Press the SPEAKER button.
2) Replace the handset on the Station.

3 Using Features

3.1 Receiving a Call

When you receive a call, your phone rings, the LCD displays the incoming call information and the associated line button flashes.

To answer a ringing call

- ✓ Lift the handset. You may press the Speaker button, line button or the Answer softkey to activate the speakerphone. To use a headset, press the Headset button.

To end the call

- ✓ Hang up the handset. If using the speakerphone, press the Speaker button. If using the headset, press the Headset button.

To refuse a ringing call

- ✓ Press Refuse softkey, the IP Phone stops ringing, and the call server routes the call.

3.2 Placing a Call

You place calls with the handset on-hook or off-hook. Similar to your mobile phone, the phone uses a technique called ‘store and forward dialing’.

After you dial a digit, the IP Phone waits about 3 seconds for additional digits. If you do not dial additional digits, the IP Phone sends the digits you dialed to the call server for processing.

To send the digits immediately, press the # button to indicate dialing is complete, and then the IP Phone sends the number to the call server.

To place a call:

1) Lift the handset. You may press the Speaker button or line button to activate the speakerphone. To use a headset, press the Headset button.
2) Dial the destination number. After a short delay, the IP Phone sends the number, or dial # to send the dialed digits immediately.
3) When the called party answers, begin speaking, the LCD displays the elapsed call time.

To end the call:

- ✓ Hang up the handset. If using the speakerphone, press the Speaker button. If using the headset, press the Headset button

3.3 Putting Call on Hold

You can place a call in a waiting state where the held party cannot hear you and you cannot hear the held party. The held party receives MOH (Music-On-Hold), if configured in the call server.

While a call is on hold, you can answer or place other calls and can toggle between held calls on the same line. With Call Hold Ringback enabled, the IP Phone rings as a reminder of calls on hold when you return the IP Phone to idle. Refer to your system administrator for further information.

To place a call on hold:

- ✓ Press the Hold button or the Hold softkey. The call status changes to hold, and you receive dial tone. You may then select another line to answer or place a call.

To retrieve a held call on a different line:

1) Press the slow flashing line button.
2) Select the desired call from the displayed call list using Previous-Next softkeys.
3) Select the OK softkey to resume a connection with the call.

To toggle between two calls on the same line:

1) Press the line button.
2) Press the up or down key of the navigation if it has navigation keys.
3) The active call status changes to hold and the held call connects.

3.4 Announcing a Call Transfer

Using Announcing a Call Transfer, you send an active call to another extension after announcing the call transfer to the receiving party.

To transfer an active call:

1) Press the Transfer button or the Transfer softkey. The active call status changes to hold and you receive dial tone.
2) Dial the number you wish to receive the transfer.
3) Await answer and announce the transfer.

To complete the transfer:

- ✓ Hang up the handset.

To cancel a transfer, for example the second party does not answer or is busy:

- ✓ Press the EndCall softkey.

3.5 Using Blind Call Transfer

You can transfer an active call to another extension without talking to the party receiving the call. The call transfers to the new party without verifying that the new party is there or willing to receive the call.

To transfer an active call without announcing the transfer:

1) Press the BlindXfr softkey. The active call status changes to hold and you receive dial tone.
2) Dial the number you wish to receive the transfer.
3) To complete the transfer, hang up the handset.

To cancel a transfer, for example you mis-dial the receiving number:

- ✓ Press the EndCall softkey, before the IP Phone sends the number.

3.6 Call Forward

Call forward directs incoming calls to ring at another location that you designate. You can select one of four (3) conditions that will forward your calls:

- **Call Forward All Calls** – immediately redirects all incoming calls.
- **Call Forward Busy** – redirects incoming calls if the line is busy.
- **Call Forward No Answe**r – redirects incoming calls that are not answered in the configured no answer time.

To set up Call Forward:

1) Press the CallFwd softkey.
2) Use the Previous-Next softkeys to select the desired Call Forward condition.
3) Press the OK softkey.
4) Input the destination where you want your calls sent.
5) Press the OK softkey. The display shows “(x) Fwd to ext” indicating the forward condition and destination for the calls.
   (U) = Unconditional, (B) = Busy, (N) = No-answer

To disable Call Forward:

1) Press the CallFwd softkey.
2) Press the Disable softkey.
3) The message “Fwd to xxxx” disappears from the LCD.

3.7 3-Party Conference

The IP Phone has a 3-Party Conference feature that lets you converse with two other parties simultaneously.

To set-up a conference:

1) Place first call as normal.
2) Press the Conferen softkey. The call status changes to hold, and you receive dial tone.
3) Dial the number you want to join the conference.
4) When the party answers, press the Join softkey to establish the conference.

To end your conference:

✓ Hang up the handset.

To cancel a conference, for example the second party does not answer or is busy:

✓ Press the EndCall softkey.

3.8 Call Waiting

With Call Waiting configured, while you are in a call the IP Phone alerts you of an incoming call. The line button flashes and you receive a beep tone over the active conversation to indicate the new call. In addition, during the ring signal, the LCD displays the incoming caller id so you know who is calling before you interrupt the current call.

To answer an incoming call while on a call:

- ✓ Press the flashing line buttonr or the Answer softkey. The active call changes to the hold state and the new call connects

3.9 Redial

Your IP Phone saves the last number you dialed and allows you to redial the number with the touch of a button. The number is stored in memory but is lost if power to the IP Phone is interrupted.

To redial a call:

1) Press the Redial softkey or Mute button. The IP Phone activates the speakerphone and places the call.
2) Lift the handset for privacy or press the Headset button to use the headset.

3.10 Speed Dial Buttons

Flexible buttons not assigned as a line are available for configuration as feature buttons including Speed dial. Pressing a Speed dial button selects your line and sends the stored number to the call server for processing. Refer to your system administrator for further information.

To place a call using Speed dial:

1) Press the desired Speed dial flexible button. The IP Phone activates the speakerphone and places the call on your line. You may select a different line button on your IP Phone prior to selecting the Speed dial button.
2) Lift the handset for privacy or press the Headset button to use the headset

3.11 Speed Number Dialing

Your Phone incorporates a Phonebook with up to 1000 records. Each record includes a name, contact information and a Speed number (000 to 999). When you dial a Speed number, the IP Phone initiates a call to the contact. Speed number dialing must be enabled. Refer to your system administrator for further information.

To place a call using a Speed number:

- ✓ Dial the Speed number (000-999) for the desired contact. After a short delay, the IP Phone displays the contact information from the Phonebook. Press the OK button to set up the call immediately. If the Phonebook has no record of the Speed number, the IP Phone sends the digits dialed to the call server for processing.

3.12 DND(Do-Not-Disturb)

When the DND (Do Not Disturb) feature is enabled and activated, the IP Phone will not notify you of an incoming call. The IP Phone does not ring, allowing you to work without interruption from an incoming call. The call server configuration determines routing of the incoming call while your phone is in DND. Refer to your system administrator for further information.

To activate DND:

- ✓ While the IP Phone is idle, press the DND button. When active, the DND button LED illuminates red, and "Do not disturb" displays in the LCD.

To deactivate DND:

- ✓ Press the illuminated DND button again.

3.13 Muting a Call

Mute toggles the microphone of the Handset, Speakerphone and Headset Off and On. When Mute is active, the connected party will not receive audio from the microphone.

To mute the microphone:

- ✓ Press the Mute button. The Mute button LED illuminated red indicating mute is active.

To turn the microphone back on:

- ✓ Press the illuminated Mute button again.

3.14 Access Your Voice Mail

When you have a message waiting, the Message LED flashes to indicate you have new messages in your mailbox. The IP Phone delivers the message waiting indication for your primary line, line button 1.

To access your voice mailbox:

1) Press the Message button. The display indicates the number of messages waiting.
2) Press Dial softkey. The IP Phone activates the speakerphone and places a call to your voice mailbox. To listen to your messages, follow the instructions for your voice mail system.

3.15 DSS/BLF Coverage

The Phone has a DSS line type that addresses the special needs associated with an Attendant or Secretarial answering position. The specific needs for these users are:

- **Direct Station Selection (DSS)** — one-touch calling and transfers to extensions
- **Busy Lamp Field (BLF)** — visual status indication of covered lines
- **Coverage** — the ability to pick up calls ringing at those extensions

The DSS line in the IP Phone uses the shared line assignment in the call server. Operation of the DSS/BLF Coverage button however is distinctly different then a shared line. For more information on assigning DSS lines, consult your system administrator.

To call an idle covered extension:

1) Lift the handset. You may press the Speaker button or the headset button.
2) Press the DSS line button. You may press the DSS line button prior to lifting the handset, which activates the speakerphone and places the call.

To transfer an active call:

1) Press the DSS line button. The IP Phone places the active call on hold and calls the extension.
2) Await answer and announce the transfer.
3) To complete the transfer, hang up the handset.

To cancel a transfer, for example the second party does not answer or is busy:

✓ Press the EndCall softkey.

To answer a call for a covered extension:

1) Lift the handset. You may press the Speaker button or press the headset button.
2) Press the flashing DSS/BLF Coverage button. You may press the DSS line button prior to lifting the handset, which activates the speakerphone and answers the call.

The LED indications of the DSS line button are:

- **Solid red:** line in use remotely
- **Fast blinking red:** Call appearance ringing for covered line
- **Slow blinking red:** Remote call hold

3.16 Using the Call Log Menu

The Phone has access to a log of all calls placed or received by the phone, including incoming calls that were not answered (missed calls). The LCD displays the call logs that you can use to place a call to the logged party. You access the call logs through the Softkey or a flexible button assigned for the Call Log function. Refer to your system administrator for further information.

To access the Call Logs menu:

1) While in an idle state, press the Call Log softkey or flexible button. The Call Log menu displays with several choices (All, Missed, Received and Placed Calls).
2) Use the Previous-Next softkeys to select the desired log and press the OK softkey or dial the digit in front of the desired selection.
3) The log will display information on the calls along with the received caller id.

To place a call using the Call Logs:

✓ Dial the digit preceding the desired log item or use the Previous-Next softkeys to highlight the desired log item and press the Dial softkey.

3.17 Using the Phonebook Menu

The Phone has an internal Phonebook with up to 1000 entries. Each record includes the contact name, contact information and Speed number. You can add, delete, edit, dial and search entries from the Phonebook Menu. You access the Phonebook Menu with the Down navigation button or Fixed button or using a flexible button assigned for the Phonebook function. Refer to your system administrator for further information.

To access the Phonebook menu:

- ✓ Press the Down navigation button or Fixed button or Phonebook flexible button, the LCD displays the first Phonebook record.

To call a contact from the Phonebook:

1) Access the Phonebook as described above.
2) Use the Previous-Next softkeys to move the cursor to the desired record or dial the digit preceding the desired record.
3) Press the Dial softkey.

To add a record to the Phonebook:

1) Access the Phonebook as described above.
2) Press the Right navigation button to display the Add softkey.
3) Select the Add softkey.
4) Enter the name, contact information, ring type and speed number for the record, selecting the OK softkey for each entry. Use the Mode softkey to enter characters. Refer to 1.10.1 Phone Entering Text, which shows the relationship between the dial pad digits and characters.

To edit a record:

1) Access the Phonebook as described above.
2) Use the Previous-Next softkeys to move the cursor to the desired record or dial the digit preceding the desired record.
3) Press the Right navigation button to display the Edit softkey.
4) Select the Edit softkey.
5) Edit the name, contact information, ring type and speed number for the record, selecting the OK softkey for each entry. Use the Mode softkey to enter characters. Refer to 1.10.1 Phone Entering Text, which shows the relationship between the dial pad digits and characters.

4 Using the LCD Configuration Menu

4.1 LCD Configuration Menu

This chapter provides the information to configure the phones using the LCD Configuration Menu. The LCD displays the configuration menu and softkey functions.

The softkeys and dial pad select menu items and input values. A configuration session begins by pressing the Menu button, which accesses the LCD Configuration Menu shown in Figure 4-1. Select the desired menu item using one of the following methods:

- ✓ Use the dial pad to input the menu item number. This is the digit displayed in front of the item or Navigate with the Previous-Next softkeys to highlight the item and select with the OK softkey.

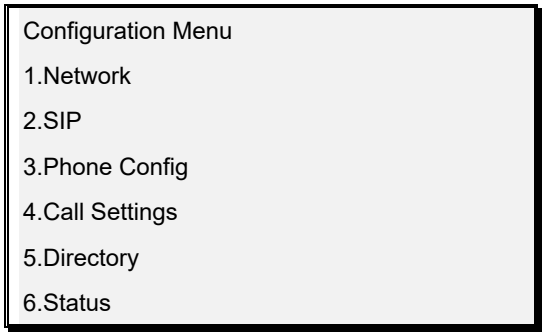


Figure 4-1: LCD Configuration Main Menu

When the desired parameter is reached, enter data with the dial pad or select values with the softkeys. Refer to Figure 4-2 (next page) for the complete menu.

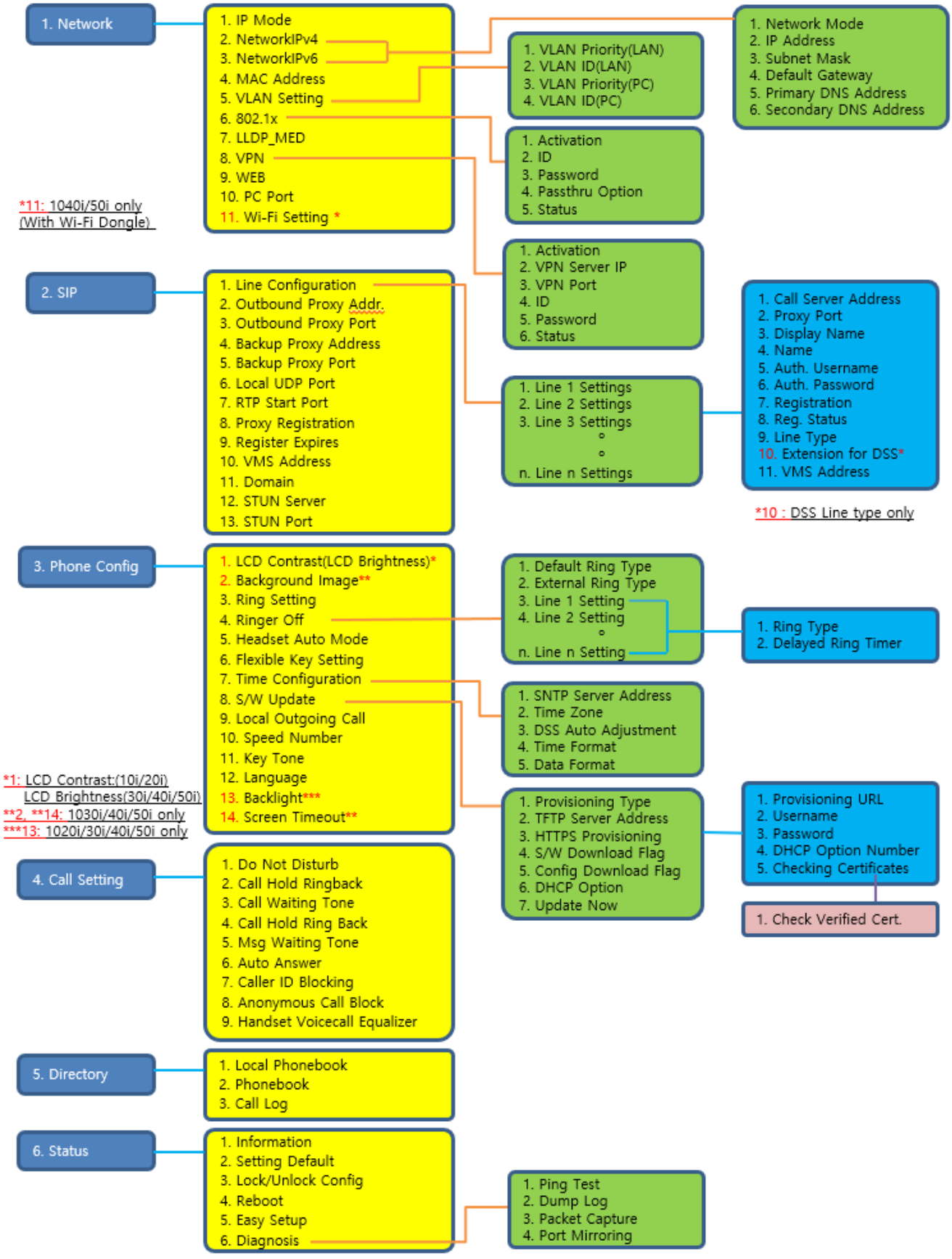


Figure 4-2: LCD Configuration Menu

4.2 Navigating the LCD Configuration Menu

The IP Phone must be idle to access the configuration menu and may be password protected (Locked). If locked, advanced configuration parameters can be viewed, however, data entry is not possible.

To access the configuration menu:

- ✓ Press the Menu button.

To select an item:

1) Dial the digit associated with the menu item or
2) Use the Previous-Next softkeys to move the cursor to the item and then press the OK softkey. The bottom line of the display shows three (3) softkey functions at a time. Additional softkey functions are available when the right arrow, →, appears in the bottom-right corner of the LCD.
3) Use the Right navigation button, to view the additional softkeys.

To return to a previous branch in the menu when the return arrow, ↰, appears in the bottom left corner of the LCD:

- ✓ Press the Left navigation button.

To exit the configuration menu:

- ✓ Press the Menu button.

4.3 Entering Values with the Dial Pad and Softkeys

Several features available to your phone require text entry. Characters are entered by pressing the dial pad digit with the character multiple times based on the character position on the button. For example, the letter ‘C’ is the third letter on the digit ‘2’ thus, to enter a ‘C’, the dial pad digit ‘2’ would be pressed 3 times.

Use the dial pad to input numbers, letters, special characters, and a period. You can enter upper or lower case letters, symbols, or numeric characters. The Mode softkey displays to indicate a parameter can accept an alphanumeric entry. Selecting the Mode softkey switches the dial pad mode. The dial pad mode displays in the upper-right corner as 1 for numeric, A for upper case characters and a for lower case characters. 1.10.1 Phone Entering Text defines the relationship between the dial pad digits and characters.

To enter characters with the dial pad, use the digit with the desired character. Press the same digit repeatedly until the desired character appears in the LCD. After entering a character, pause to allow the cursor to move to next character position.

Other special entries include:

- ✓ To enter a period or dot (for example, to input an IP address), press asterisk, ✱, on the dial pad.
- ✓ To delete a letter in front of the cursor, press the ← softkey
- ✓ To clear an entire string, press the Clear softkey.

Prior to saving an entry, you may discard changes by pressing the Left navigation button to move to the previous menu, or press the Menu button to exit the Configuration menu.

4.4 Lock/Unlock Configuration

The LCD Configuration Menu can be protected for security. With a password assigned, exiting the LCD Menu automatically locks the menu. Configuration data displays while the menu is locked however, the menu must be unlocked using the password in order to change advanced parameters. Parameters under the 'Phone Config' item listed below are not subject to the lock feature, allowing the user access without the need to enter a password.

- LCD Contrast (1010i, 1020i), LCD Brightness (1030i, 1040i, 1050i)
- Ring Type
- Ringer Off
- Headset Auto Mode
- Flexible Button settings
- Time Configuration
- Speed Number Dialing
- Key Tone
- Language
- Backlight (except of 1010i)

4.5 Network Configuration

Under Network Configuration are the Network mode (DHCP or static), various IP addresses, DNS (Domain Name Service), and other LAN parameters. In addition, you can see the MAC address of the IP Phone.

NOTE

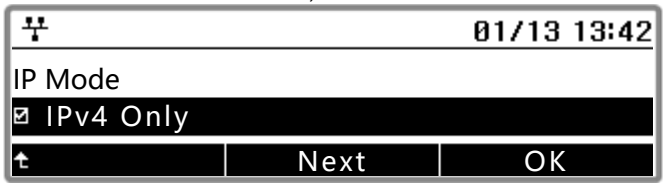
Changing network parameters requires a reboot of the IP Phone; when you exit the LCD Configuration Menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.1 IP Mode

IP Phone supports IPv4 and IPv6 both. To use only IPv4, select the "IPv4 only" from this menu. The phone will set only IPv4 to the network interface. To use only IPv6, select the "IPv6 only", and to use both IPv4 and IPv6, select the "IPv4, IPv6 Both".

To set the IP Mode:

1) Press the Menu button.
2) Select "Network" menu, then select "IP Mode" menu.



3) Select the IP mode among "IPv4 only", "IPv6 only" or "IPv4, IPv6 Both" using the navigation key and press the OK softkey to save your selection.

4.5.2 Network Mode

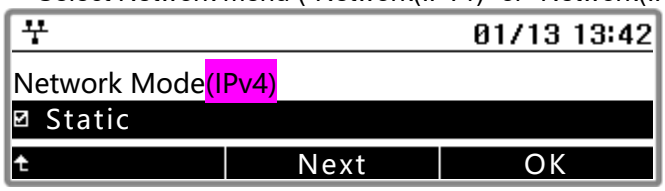
IP Phones can obtain all IP configuration data from a DHCP server. If no DHCP server is available, static addressing is used for Network Mode. To choose static addressing, you need to manually enter other IP network parameters, including:

- IP address of the phone
- Default gateway IP address
- Subnet mask
- DNS address

This setting can be set for each IPv4 and IPv6

To set the Network Mode:

1) Press the Menu button.
2) Select Network menu (“Network(IPv4)” or “Network(IPv6)”), then select “Network Mode” menu.



3) Select the Network Mode using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save your selection.

NOTE

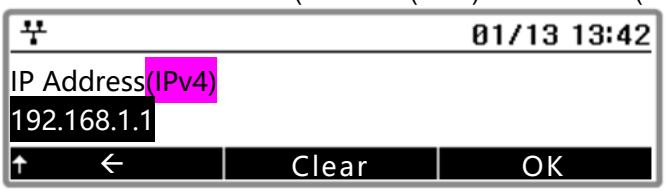
The ☑ mark displays in front of the current selection. Changing the Network Mode requires a reboot of the IP Phone; when you exit the LCD Configuration Menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered. The “(IPv4)” or “(IPv6”) is shown at the end of the title.

4.5.3 IP Address

Selecting static addressing for the Network mode requires manual entry of an IP address for the IP Phone. The IP Phone requires a valid available IP address for proper operation.

To set the Network Mode:

1) Press the Menu button.
2) Select Network menu (“Network(IPv4)” or “Network(IPv6)”), then select “IP Address” menu.



3) Input the IP Address using the dial pad and press the OK softkey to save your entry.

NOTE

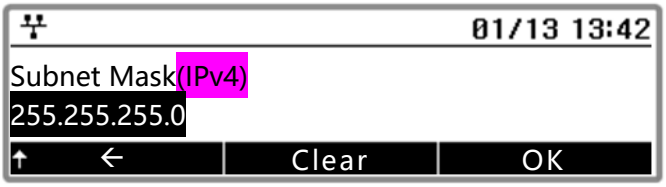
- ✓ The IP Address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP Phone retrieves an IP address from the DHCP server.
- ✓ Changing the IP Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ The “(IPv4)” or “(IPv6”) is shown at the end of the title.

4.5.4 Subnet Mask

Selecting static addressing for the Network mode requires manual entry of a subnet mask for the IP Phone. The IP Phone requires a valid subnet for proper operation.

To enter the Subnet Mask:

1) Press the Menu button.
2) Select Network menu ("Network(IPv4)" or "Network(IPv6)"), then select "Subnet Mask" menu.



3) Input the Subnet Mask using the dial pad and press the OK softkey to save your entry.

NOTE

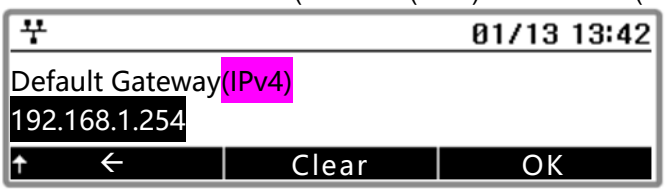
- ✓ The Subnet Mask can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP Phone retrieves a subnet mask from the DHCP server.
- ✓ Changing the Subnet Mask requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ The "(IPv4)" or "(IPv6") is shown at the end of the title.

4.5.5 Default Gateway

Selecting static addressing for the Network mode requires manual entry of a default gateway address for the IP Phone. The IP Phone requires a valid default gateway address for proper operation.

To assign the Default Gateway address:

1) Press the Menu button.
2) Select Network menu ("Network(IPv4)" or "Network(IPv6)"), then select "Default Gateway" menu.



3) Input the Default Gateway IP address using the dial pad and press the OK softkey to save your entry.

NOTE

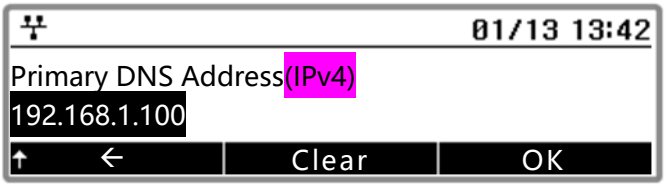
- ✓ The Default Gateway address can be changed only in static mode and not if DHCP mode is active. In DHCP mode, the IP Phone retrieves a default gateway address from the DHCP server.
- ✓ Changing the Default Gateway address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ The "(IPv4)" or "(IPv6") is shown at the end of the title.

4.5.6 Primary DNS Address

Using an FQDN (Fully Qualified Domain Name) for any of the servers in the IP Phone configuration requires entry of a primary DNS IP address. The IP Phone contacts the primary DNS server to resolve the FQDN to an IP address.

To assign the Primary DNS Address:

1) Press the Menu button.
2) Select Network menu ("Network(IPv4)" or "Network(IPv6)"), then select "Primary DNS Address" menu.



3) Input the Primary DNS address using the dial pad and press the OK softkey to save your entry.

NOTE

Changing the Primary DNS Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered. The "(IPv4)" or "(IPv6") is shown at the end of the title.

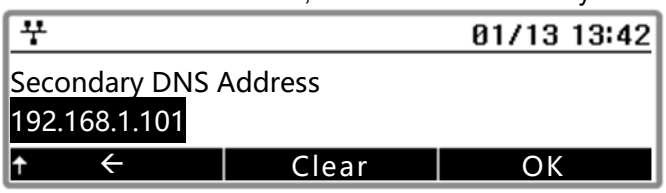
4.5.7 Secondary DNS Address

With an FQDN assigned as any of the servers in the IP Phone configuration, the IP Phone contacts the primary DNS server to resolve the FQDN to an IP address. Should the primary not respond, the IP Phone contacts a secondary DNS for FQDN resolution.

This is an optional but recommended setting.

To assign the Secondary DNS Address:

1) Press the Menu button.
2) Select "Network" menu, then select "Secondary DNS Address" menu.



3) Input the Secondary DNS address using the dial pad and press the OK softkey to save your entry.

NOTE

Changing the Secondary DNS Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

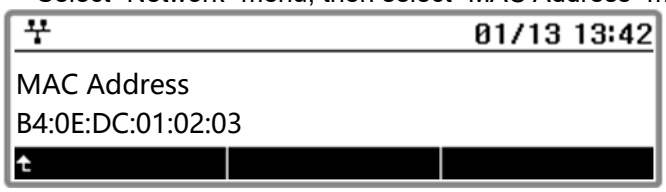
4.5.8 MAC Address

IEEE specifications require manufacturers of data equipment to store a unique 6-byte MAC or Ethernet address in the permanent memory of each network device.

To view the MAC Address:

1) Press the Menu button.

2) Select “Network” menu, then select “MAC Address” menu.



3) View the MAC address of the IP Phone.

4.5.9 VLAN Settings

VLAN settings define the Ethernet frame priority and VLAN identification in accordance with IEEE 802.1p/Q standards.

Separate VLAN tags are assigned for the IP Phone PC port and the LAN (voice) port.

With a VLAN ID assigned, only frames with the assigned ID are accepted, and all SIP and voice frames sent by the phone include the VLAN ID assigned to the LAN port.

NOTE

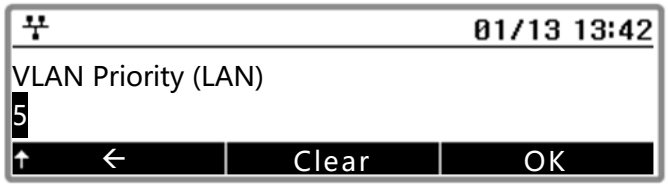
- ✓ For proper operation, other network elements (LAN switches and default gateway) must support and be configured with proper VLAN parameters. VLAN settings can be overwritten if LLDP is enabled and the VLAN policy is LLDP.
- ✓ Changing the VLAN parameters requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.9.1 VLAN Priority (LAN)

The VLAN Priority (LAN) establishes the priority for Ethernet frames from the phone voice port including signaling and voice packets. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign VLAN Priority for the LAN (voice) port:

1) Press the Menu button.
2) Select “Network” menu.
3) Select “VLAN Settings” menu, then select “VLAN Priority (LAN)” menu.



4) Input the VLAN Priority Value for the LAN port using the dial pad and press the OK softkey to save your entry.

NOTE

- ✓ Changing the VLAN Priority requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ The recommended value for VLAN Priority (LAN) port is 5.
- ✓ To disable VLANs, assign the VLAN ID as zero (0).

4.5.9.2 VLAN ID (LAN)

The VLAN ID (LAN) assigns the identification for Ethernet frames for the IP Phone voice port including signaling and voice payloads.

Setting the VLAN ID to zero (0) disables the VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN ID for the LAN (voice) port:

1) Press the Menu button.
2) Select “Network” menu.
3) Select “VLAN Settings” menu, then select “VLAN Priority (PC)” menu.

4) Input the VLAN Priority for the PC port using the dial pad and press the OK softkey to save your entry.

NOTE

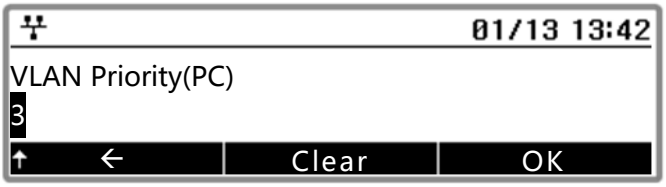
- ✓ Changing the VLAN ID requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ To disable VLANs, assign the VLAN ID as zero (0).

4.5.9.3 VLAN Priority (PC)

The VLAN Priority (PC) establishes the priority for Ethernet frames for the IP Phone PC port. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN Priority for the PC port:

1) Press the Menu button.
2) Select “Network” menu.
3) Select “VLAN Settings” menu, then select “VLAN Priority (PC)” menu



4) Input the VLAN Priority for the PC port using the dial pad and press the OK softkey to save your entry.

NOTE

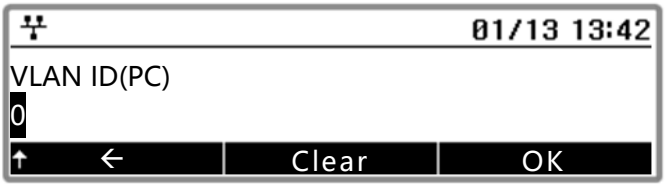
- ✓ Changing the VLAN Priority for the PC port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ The recommended value for VLAN Priority (PC) port is 3.
- ✓ To disable VLAN for the PC port, assign the VLAN ID (PC) as zero (0).

4.5.9.4 VLAN ID (PC)

The VLAN ID (PC) assigns the identification for Ethernet frames for the IP Phone PC port. Setting the VLAN ID to zero (0) disables VLAN framing and the IP Phone uses only standard Ethernet frames.

To assign the VLAN ID (PC) port:

1) Press the Menu button and select "Network" menu.
2) Select "VLAN Settings" menu, then select "VLAN ID (PC)" menu.



3) Input the VLAN ID for the PC port using the dial pad and press the OK softkey to save your entry.

NOTE

- ✓ Changing the VLAN ID (PC) requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.
- ✓ To disable VLAN (PC), assign the VLAN ID (PC) as zero (0).

4.5.10 802.1x

The 802.1x protocol is an IEEE standard for media-level access control offering the capability to permit or deny network connectivity, control LAN access, and apply traffic policy, based on user or machine identity.

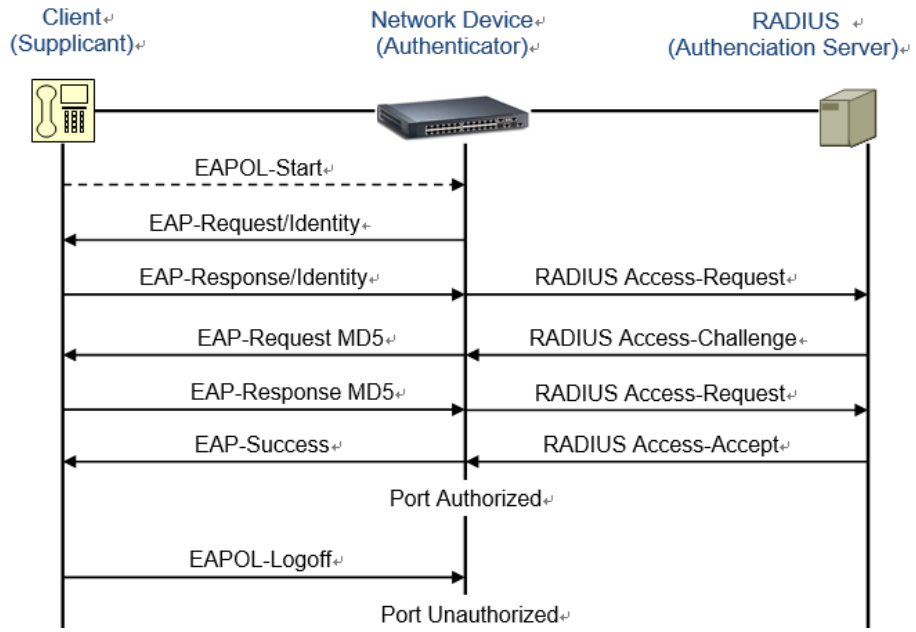
The 802.1x protocol consists of three components (or entities):

- Supplicant – a port access entity (PAE) that requests access to the network. For example, IP Phone and the attached PC can be 802.1X supplicants.
- Authenticator – a PAE that facilities the authentication of the supplicant. For example, the switch as an authenticator PAE that controls the physical access to the network based on the authentication status of a supplicant.
- Authentication server – a PAE, usually a Remote Authentication Dial-In User Service (RADIUS) server that provides the authentication service.

The 802.1X protocol makes use of Extensible Authentication Protocol (EAP) messages. The protocol in 802.1X is called EAP encapsulation over LANs (EAPOL). The Authenticator becomes the middleman for relaying EAP received in 802.1X packets to an authentication server by using the RADIUS format to carry the EAP information

1) Supplicant sends EAPOL-Start frame to Authenticator.
2) Authenticator sends EAP-Request/Identity packet to Supplicant.
3) Supplicant sends EAP-Response/Identity packet includes User ID to Authenticator.
4) Authenticator sends it as a RADIUS Access-Request packet to Authentication Server.
5) Authentication Server sends RADIUS Access-Challenge packet to Authenticator to decide the EAP Method.
6) If Authentication Server and Supplicant agree with EAP Method, exchange EAP Request and Response between Authentication Server and Supplicant through the Authenticator.

7) Authentication Server sends EAP-Success (RADIUS Access-Accept) or EAP-Fail (RADIUS Access-Reject.).



NOTE

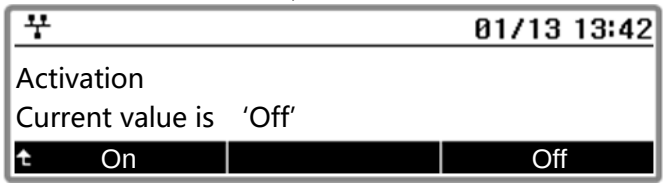
Changing the 802.1x parameters requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.10.1 Activation

It determines whether the IP Phone responds as a supplicant.

To set the supplicant option:

1) Press the Menu button and select “Network” menu.
2) Select “802.1x” menu, then select “Activation” menu.



3) Select the softkey (On or Off) to enable or disable the supplicant function.

NOTE

Changing the supplicant option requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.10.2 ID

The IP Phone is challenged to access the network by specifying a username and password when supplicant option is enabled.

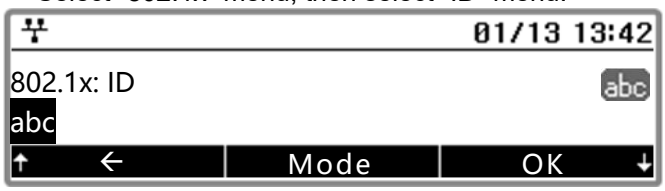
Authentication usernames and passwords for each supplicant device must be provisioned in the RADIUS server. The default username is empty.

To assign supplicant ID:

1) Press the Menu button.

2) Select “Network” menu.

3) Select “802.1x” menu, then select “ID” menu.



4) Input the supplicant ID using the dial pad and press the OK softkey to save your entry.

NOTE

✓ Changing the supplicant ID requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.10.3 Password

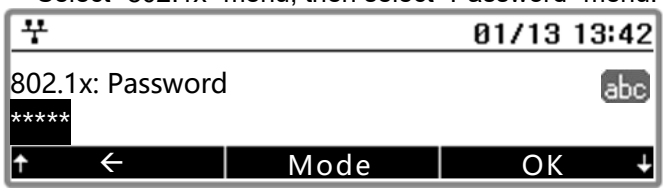
The IP Phone is challenged to access the network by specifying a username and password when supplicant option is enabled. Authentication usernames and passwords for each supplicant device must be provisioned in the RADIUS server. The default password is empty.

To assign supplicant ID:

1) Press the Menu button.

2) Select “Network” menu.

3) Select “802.1x” menu, then select “Password” menu.



4) Input the supplicant password using the dial pad and press the OK softkey to save your entry.

NOTE

✓ Changing the supplicant password requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.10.4 PassThru Mode

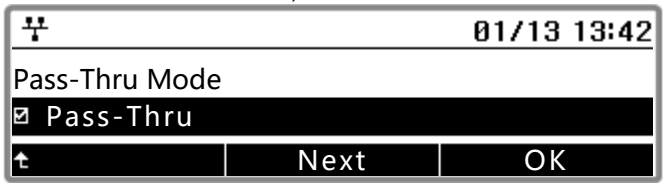
The PassThru mode for the 802.1x can be set by this menu. Possible values are “Pass-Thru”, “Pass-Thru Off”, or “Pass-Thru with LogOff”.

To assign the PassThru mode:

1) Press the Menu button.

2) Select “Network” menu.

3) Select “802.1x” menu, then select “PassThru Mode” menu.



4) Select the Pass-Thru mode using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save your selection.

NOTE

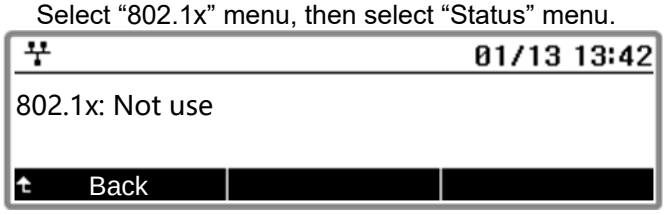
Changing the Supplicant Pass-Thru mode requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.10.5 Status

User can check the status as a supplicant of the IP Phone.

To view 802.1x status:

1) Press the Menu button.
2) Select “Network” menu.
3) Select “802.1x” menu, then select “Status” menu.



4) See the current status of the 802.1x.

4.5.11 LLDP-MED

The IEEE 802.1AB Link Layer Discovery Protocol (LLDP) defines a standard method for LAN devices to inform each other about their configurations. The 802.1AB standard defines a set of advertisement messages, called type-length-values (TLVs).

LLDP Media Endpoint Discovery (LLDP-MED) protocol is an enhancement to the 802.1AB standard that provides “plug and play” capability for VoIP networks.

The 1000i phones supports for LLDP and LLDP-MED extensions provides the ability to use discovered information such as device type, software version and serial number, and other information for inventory management.

Through the LLDP protocol, the 1000i phones communicates with the switch to learn the voice VLAN ID. There is no longer a need to configure VLAN tagging manually.

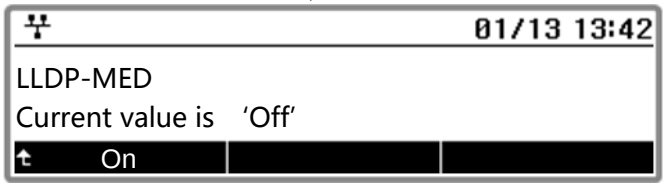
The following is a list of supported TLVs:

No	Name	Information
1	Chassis-Id	IP address of phone (4 bytes) Note: 0.0.0.0 is sent until the phone has a valid IP address.
2	Port-Id	MAC address of phone (6 bytes)
3	Time-To-Live	Nmber of seconds that the recipient should consider the LDP information to be valid, 120 seconds.
4	System Name	iPECS IP98xx
5	System Description	Includes Manufacturer′s name, Model name, H/W version, Application version and BootROM version
6	Capabilities	Telephone and Bridge if the phone has PC port support
7	Management Address	IP address of phone
8	LLDP-MED Capabilities	Identifies the types of LLDP MED capabilities supported by the phone. Capabilities - 0x33 (LLDP-Med capabilities, Network policy, Extended Power Via MDI-PD, Inventory), Class Type III

9	Network Policy	Specifies the VLAN ID, the 802.1 priority, and the differentiated-services-code-point (DSCP) value ApplicationType: Voice (1) Policy: (Unknown(=1)/Defined(=0) Unknown, if phone is in booting stage or if switch doesn't support network policy TLV. Defined, if phone is operational stage and Network policy TLV is received from the switch.), Tagged/Untagged, VlanId, L2 priority and DSCP
10	Extended Power Via-MDI	Contains information related to how the device is powered, power proirity, and how much power the device needs Refer to the Model Name & Power Values Table.
11	LLDP-MED inventory - Hardware revision - Firmware revision - Software revision - Serial number - manufacturer name - Model name	 - H/W version of phone - BootROM version of phone - Application version of phone - Mac address of phone (ASCII string) - iPECS - Model name of phone: Refer to the Model Name & Power Values Table.
12	End of LLDPDU	Indicates the end of an LLDP data unit

To set the LLDP-MED option:

1) Press the Menu button.
2) Select “Network” menu, then select “LLDP-MED” menu.



3) Select the desired softkey (On or Off) to enable or disable LLDP-MED function.

NOTE

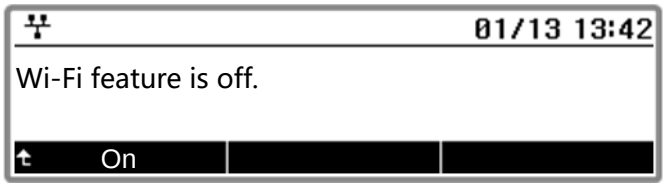
Changing the LLDP-MED option requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.5.12 Wi-Fi Settings

The 1040i and 1050i support the Wi-Fi dongle. If a Wi-Fi dongle is attached to the 1040i or 1050i, you can see an addition menu, “Wi-Fi Settings”.

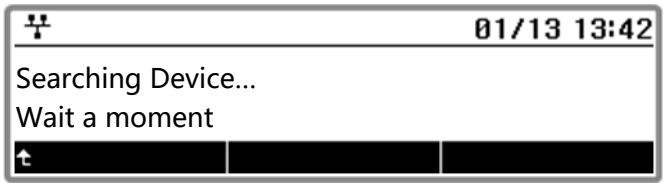
4.5.12.1 Activation of the Wi-Fi dongle

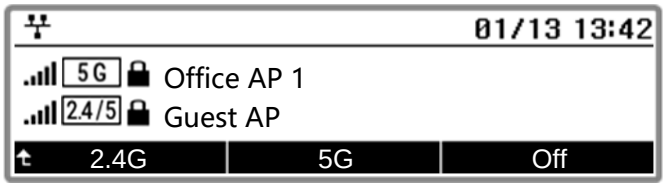
At the first time after the Wi-Fi dongle is attached, the Wi-Fi feature is not yet activated. If you see the below screen when you select the “Wi-Fi Settings” menu, press the “On” soft key to activate the Wi-Fi dongle.



4.5.12.2 Scan available APs and connect

If any AP is not connected to your Wi-Fi dongle yet, the phone will search some available APs and show the scanned APs list. If you want to see only 2.4G APs or 5G APs, press the “2.4G” or “5G” soft key. If you press the “2.4G/5G” soft key, all of APs will be shown.



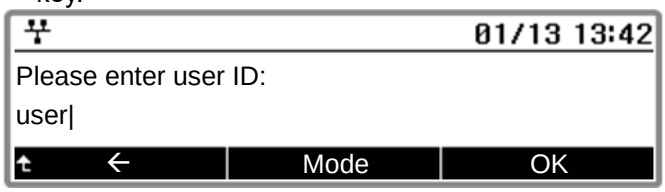


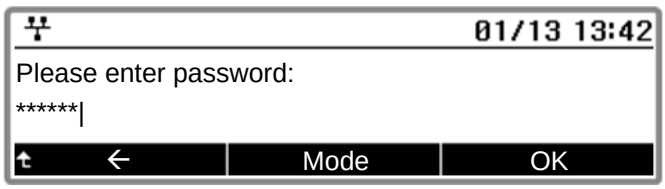
NOTE

The head of each AP list represents the status of AP. The first, . .ıl .ıl .ıl .ıll represents the strength of signal. The second, 2.4G 5G 2.4/5 shows the band of AP. If an AP is secured, the 🔒 will be shown. If it is a saved AP, the 🔖 will be shown. And it is already connected to the phone, the 📡 will be shown.

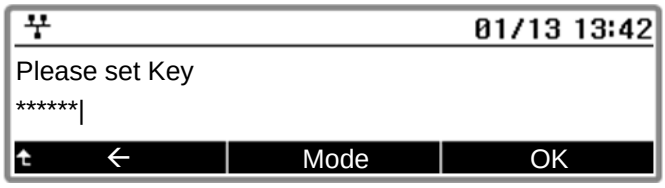
To connect an AP:

1) Select an AP using Next - Previous softkeys or VOL ▲▼ button.
2) Press the Connect soft key.
3) If the selected AP is not secured, it will be connected directlry. But, if it is secured, you should set the authentication. According to AP’s secure setting, you should enter an ID and password, or a key.

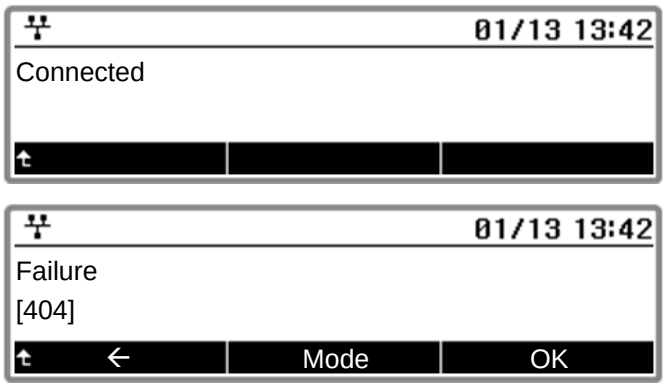




Or



4) If AP is connected successfully, you can see the first screen. But if you failed, you will see the second screen with an error code. Please check your authentications or network environment according to the error code.

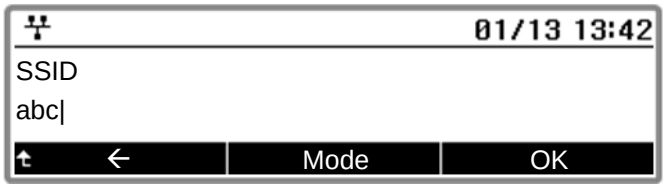


NOTE

After connecting to an AP, you should set the network configurations (network mode, IP, subnet-mask, default gateway, …) from the each menu. And you should reboot the phone after setting completely.

If you try to connect a saved AP, the phone will not ask ID & password or key because it has the previous values. If you want to connect using a new authentications, press the “Edit” soft key after selecting a saved AP. The phone will ask new authetications to you.

If you connect manually to an AP that not scanned, or a hidden AP, press the “Add” soft key. You can enter the SSID of AP and the authetications.



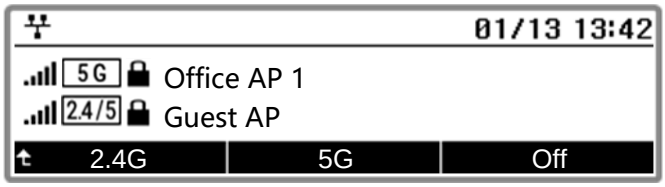
4.5.12.3 Check status of connected AP

If Wi-Fi dongle is already connected to an AP, the status of the connected AP will be shown when you enter the “Wi-Fi Settings” menu. If you press the “Disconn” soft key, the phone will disconnect from the connected AP and scan APs. If you press the “Delete” soft key, the phone will disconnect from the connected AP and delete the saved authetications of AP. If you want to connect any other AP, press the “Find” soft key. The phone will scan APs.

4.5.12.4 Deactivation of the Wi-Fi dongle

If you don’t want to use the Wi-Fi dongle any more, you can deactivate the Wi-Fi dongle by pressing the “Off” soft key from the below screens.

(from the status screen)



(from the scan list screen)

4.6 SIP Configuration

The SIP Configuration selection establishes parameters for each line appearance, SIP User Id, allowing the IP Phone to register and, if required, subscribe for the line appearance with the appropriate call server. Additionally, general Voice over IP (VoIP) characteristics are assigned, which affect characteristics associated with all lines.

Lines are assigned to the IP Phone flexible buttons in consecutive order from the first button. For each line, the following parameters are available:

- call server IP address
- authorization Id and password
- line type and special line attributes

As a default, the number of lines is one and the first line appearances are available using the LCD Menu. When the IP Phone requires multiple lines, use the Web Manager or configuration file to change the number of lines and line parameters

NOTE

For proper operation, assign the first line button as an appearance of the user’s line. For features where the IP Phone uses one-button line access such as Speed Dial, the IP Phone employs the first line.

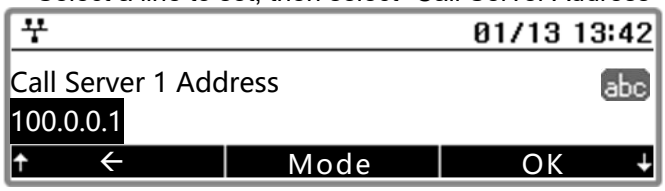
4.6.1 SIP Configuration, Line Parameters

4.6.1.1 Call Server Address

The Call Server Address is the IP address or FQDN of the IP Phone host call server. Different call servers can be defined for each line.

To assign the Call Server address:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Call Server Address” menu.



4) Input the Call Server Address or FQDN using the dial pad and press the OK softkey to save your entry.

NOTE

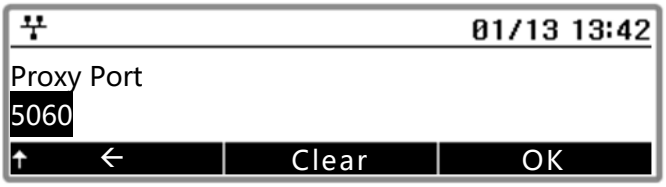
- ✓ Changing the Call Server Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.1.2 Proxy Port

The Proxy Port defines the TCP/UDP port number employed for SIP signaling transport. The port number “5060” is usually used.

To change the proxy port number:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Proxy Port” menu.



4) Input the Proxy Port number for the PC port using the dial pad and press the OK softkey to save your entry.

NOTE

Changing the Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

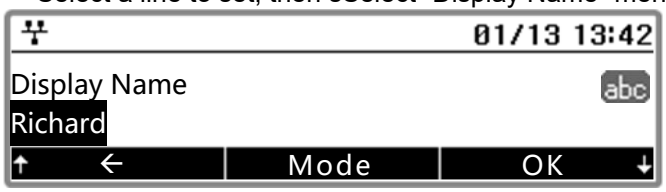
4.6.1.3 Display Name

When assigned, Display Name is used in SIP headers as the caller id name.

To input the display name:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.

3) Select a line to set, then sSelect “Display Name” menu.



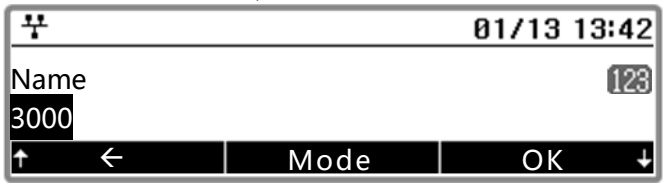
4) Input the Display Name using the dial pad and press the OK softkey to save your entry.

4.6.1.4 Name

Name is used in SIP headers as the SIP user id or account assigned in the call server. A name is required for proper operation and must match the user id or account assigned in the call server.

To input the name:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Name” menu.



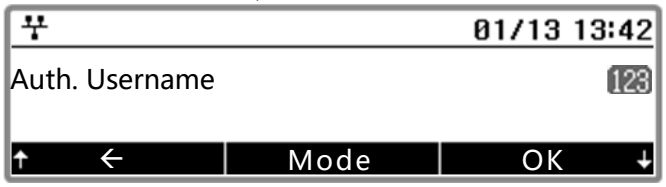
4) Input the Name using the dial pad and press the OK softkey to save your entry.

4.6.1.5 Authentication Username

Authentication of the line appearance with the SIP call server uses Authentication User Name.

To assign Authentication Username:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Auth. Username” menu.



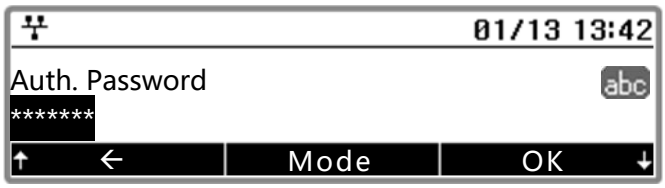
4) Input the Authentication User Name using the dial pad and press the OK softkey to save your entry

4.6.1.6 Authentication Password

Authentication of the line appearance with the SIP call server uses the Authentication Password in conjunction with the user name.

To assign the authentication password:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Auth. Password” menu.



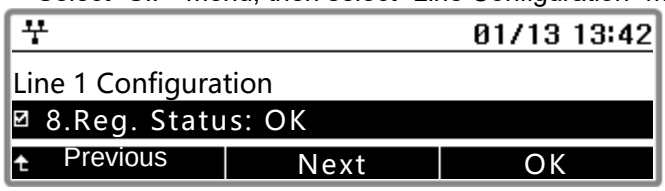
4) Input the Authentication Password using the dial pad and press the OK softkey to save your entry.

4.6.1.7 Registration Status

The status of the line registration with the SIP call server can be viewed. The status is displayed as OK, NOK (Not OK) or undefined. The latter displays when no call server address is assigned.

To view the registration status for a line:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.



3) Select a line to set, then select “Reg. Status” menu
4) See the current state.

4.6.1.8 Line Type

Each line button represents a SIP user id or account in the call server. A line must have a Line Type assignment. The three line types available are private, shared or BLA (Bridged Line Appearance) and DSS (Direct Station Selection).

A private line generally appears on line buttons of one phone but may appear at multiple phones as a Multiple Line Appearance (MLA). While incoming calls on a private line ring at all phones with an appearance, an active or held private line call is accessible only to the associated phone.

In addition, the status of an active or held private line call displays only at the associated phone. Only the active user may transfer or conference calls on a private line.

A BLA appears at line buttons of more than one phone. All phones with an appearance can access incoming and held calls on the BLA and receive status information for the call. Note for proper operation, assign the line as a BLA in the call server as well as the IP Phone. A DSS line incorporates three functions associated with the needs of an answering position:

- DSS – while the line is idle, the button calls the associated extension.
- BLF– the Busy Lamp Field LEDs of the line button display the line status.
- Call Coverage – incoming calls on the line are answered using the button.

Assign a DSS line as a BLA or BLF in the call server and as a DSS line in the IP Phone. When using the BLF function, the DSS Line type is assigned in the IP Phone. Also, for the BLF function, a URL must be defined.

Each line button employs LEDs to indicate status of the line as shown below.

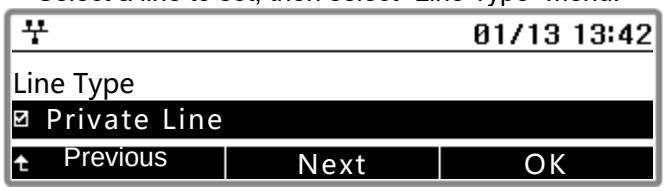
Call Status	Color	LED Status
Incoming call	Orange*	LED flashes at 60 ipm, fast flash
Line in use at this phone	Orange*	LED On

Call held at this phone	Orange*	LED flashes at 30 ipm, slow flash
BLA/DSS in use at another phone	Red	LED On
BLA/DSS Call held at another phone	Red	LED slow flash
DSS incoming call	Red	LED flashes at 60 ipm, fast flash
Line not registered w/server	Red / Orange	LED flashes color, red to orange at 30 ipm

* When the LCD displays information for a line, such as when an incoming call first rings, the LED color is orange otherwise, the LED color will be green.

To assign the Line Type:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Line Type” menu.



4) Select the type for the line menu to save your selection.

NOTE

Changing the Line Type requires a reboot of the IP Phone; when you leave the menu, you will receive a Reboot notice.

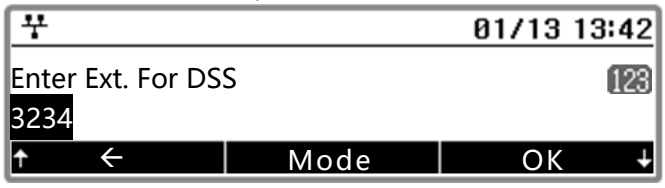
The IP Phone must reboot to utilize the new values entered.The ☑ mark displays in front of the current selection.

4.6.1.9 Extension for DSS Line

A DSS Line requires entry of the associated extension number. Note do not assign a DSS extension number for a private or shared line type.

To assign an extension number for a DSS line:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “Extension for DSS” menu.



4) Input the extension number for the DSS line using the dial pad.
5) Press the OK softkey to save your entry.

NOTE

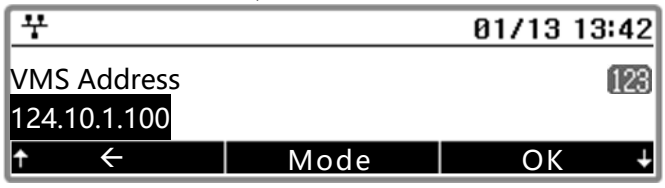
- ✓ Assign an extension number only for a DSS line.
- ✓ Do not assign an extension number for a private or shared line.

4.6.1.10 VMS Address

This menu set the dial number to connect to the VMS.

To assign a VMS address for a line:

1) Press the Menu button.
2) Select “SIP” menu, then select “Line Configuration” menu.
3) Select a line to set, then select “VMS Address” menu.



4) Input the VMS address using the dial pad.
5) Press the OK softkey to save your entry.

4.6.2 SIP Configuration, General Parameters

Parameters in this section assign the overall SIP operation.

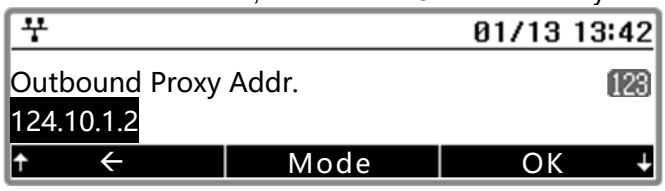
4.6.2.1 Outbound Proxy Address

With the Outbound Proxy Address defined, the IP Phone sends all requests to the proxy instead of the SIP call server configured in the section 4.6.1.1

The address may be in the form of an IP address or a FQDN.

To assign an Outbound Proxy Address:

1) Press the Menu button.
2) Select “SIP” menu, then select “Outbound Proxy Addr.” Menu.



3) Input the Outbound Proxy Address using the dial pad and press the OK softkey to save your entry.

NOTE

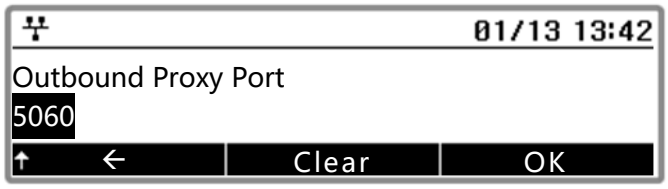
Changing the Outbound Proxy Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.2 Outbound Proxy Port

With the Outbound Proxy Server defined, the associated port must be defined as the Outbound Proxy Port. All SIP requests are sent to the assigned port instead of the proxy port configured in the section 4.6.1.2

To assign the Outbound Proxy Port:

1) Press the Menu button.
2) Select “SIP” menu, then select “Outbound Proxy Port” Menu.



3) Input the Outbound Proxy Port using the dial pad.
4) Press the OK softkey to save your entry.

NOTE

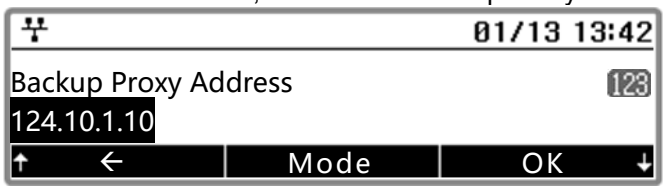
Changing the Outbound Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.3 Backup Proxy Address

When employing an outbound proxy, a backup proxy can be defined for use should the primary proxy fail to respond. The Backup Proxy Address is an IP address or a FQDN.

To assign a Backup Proxy Address:

1) Press the Menu button.
2) Select “SIP” menu, then select “Backup Proxy Address” Menu.



3) Input the Backup Proxy Address using the dial pad.
4) Press the OK softkey to save your entry.

NOTE

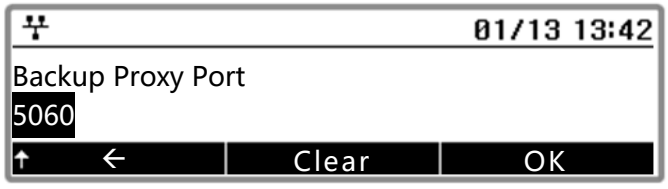
Changing the Outbound Proxy Address requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.4 Backup Proxy Port

The Backup Proxy Port defines the port used when sending SIP messages to the backup proxy defined under the section 4.6.2.3

To assign the Backup Proxy Port:

1) Press the Menu button.
2) Select “SIP” menu, then select “Backup Proxy Port” Menu.



3) Input the Backup Proxy Port using the dial pad and press the OK softkey to save your entry.

NOTE

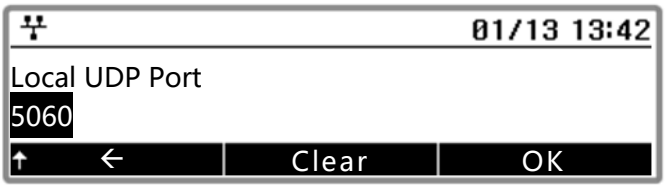
Changing the Backup Proxy Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.5 Local UDP Port

The Local UDP Port defines the port, normally 5060, the IP Phone uses to send and receive SIP signaling packets. In some instances, particularly when behind a firewall, the normal port may not be available. If port 5060 is not available, a different port can be defined.

To set the Local UDP Port:

1) Press the Menu button.
2) Select “SIP” menu, then select “Local UDP Port” Menu.



3) Input the Local UDP Port using the dial pad and press the OK softkey to save your entry.

NOTE

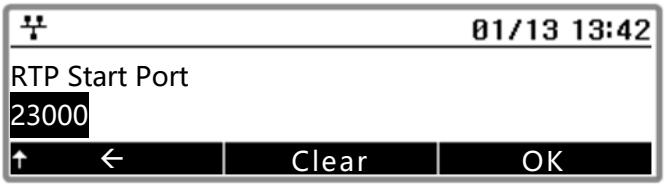
Changing the Local UDP Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.6 RTP Start Port

When a SIP call is established, Real-Time Transport Protocol (RTP) packets transport media and digitized voice. The port used for RTP packets is usually 23000, but a different port may be defined.

To assign an RTP Start Port:

1) Press the Menu button.
2) Select “SIP” menu, then select “RTP Start Port” Menu.



3) Input the Local UDP Port using the dial pad and press the OK softkey to save your entry.

NOTE

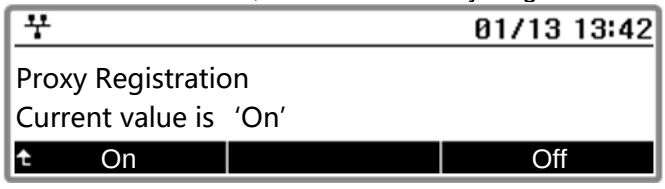
Changing the RTP Start Port requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

4.6.2.7 Proxy Registration

If Proxy Registration is set to [Yes], the IP Phone will register with the appropriate call or proxy server upon initial power-up, during a reboot and at periodic intervals based on the timer defined in the section 4.6.2.8

To enable Proxy Registration:

1) Press the Menu button.
2) Select “SIP” menu, then select “Proxy Registration” Menu.



3) To enable registration, press the On softkey or, to disable SIP registration, press the Off softkey.

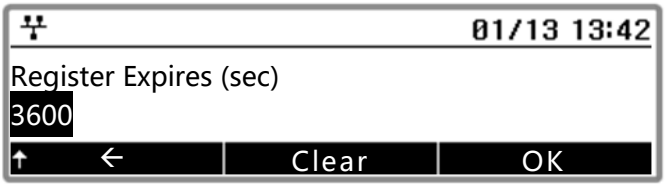
4.6.2.8 Register Expires

To assure a connection with the proxy or SIP call server, the IP Phone periodically re-registers.

Unless defined by the host server during registration, the Register Expires timer determines the interval between registration attempts and is set from 0 to 6400 seconds, default 3600 seconds.

To modify the Register Expires timer:

1) Press the Menu button.
2) Select “SIP” menu, then select “Register Expires” Menu.



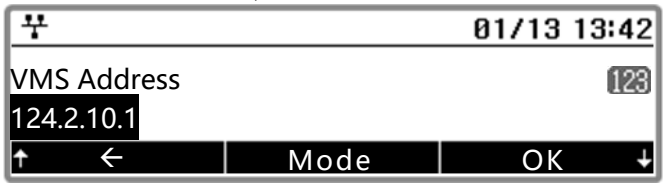
3) Input the registration expiration time using the dial pad and press the OK softkey to save your entry.

4.6.2.9 VMS Address

The Voice Mail Server (VMS) address is the IP address or phone number of the user’s voice mail system.

To assign the VMS Address:

1) Press the Menu button.
2) Select “SIP” menu, then select “VMS Address” Menu.



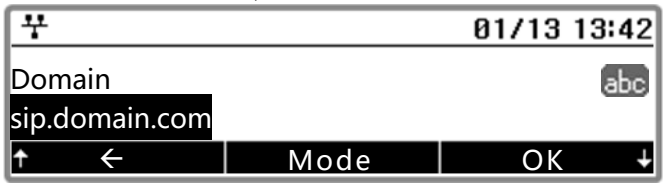
3) Input the VMS Address using the dial pad and press the OK softkey to save your entry.

4.6.2.10 Domain

The Domain specifies the domain for SIP service when the provider desires to use a specific SIP domain for the IP Phone.

To assign the Domain:

1) Press the Menu button.
2) Select “SIP” menu, then select “Domain” Menu.



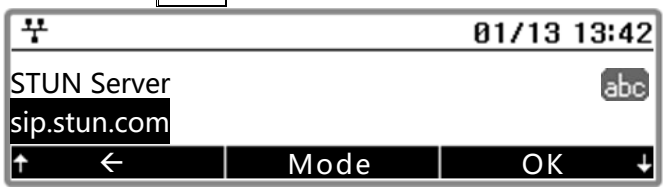
3) Input the Domain using the dial pad and press the OK softkey to save your entry.

4.6.2.11 STUN Server

The STUN server is for Session Traversal Utilities for NAT(STUN).

To assign the STUN Server:

1) Press the Menu button.



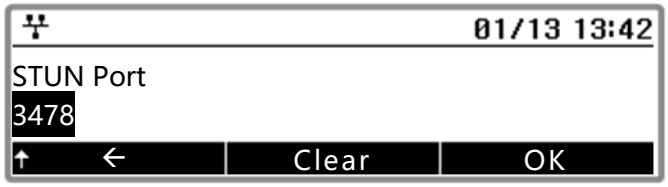
2) Select “SIP” menu, then select “STUN Server” Menu.
3) Input the STUN server using the dial pad and press the OK softkey to save your entry.

4.6.2.12 STUN Port

The STUN port defines the port for STUN Server.

To assign the STUN Port:

1) Press the Menu button.
2) Select “SIP” menu, then select “STUN Port” Menu.



3) Input the STUN Port using the dial pad and press the OK softkey to save your entry.

4.7 Phone Settings

Under the Phone Settings menu, the user parameters listed below can be changed.

- LCD contrast (1010i, 1020i)
- LCD brightness (1030i, 1040i, 1050i)
- Ring setting
- Ringer Off
- Headset auto mode
- Flexible button settings
- Time Configuration
- S/W Update
- Outgoing Call Lock
- Speed Number Dialing
- Key Tone
- Language
- Backlight (1010i, 1020i)
- Screen Timeout (1030i, 1040i, 1050i)

With the exception of S/W Update, these settings are not password protected. The user may access these parameters even if the LCD Configuration Menu is locked.

NOTE

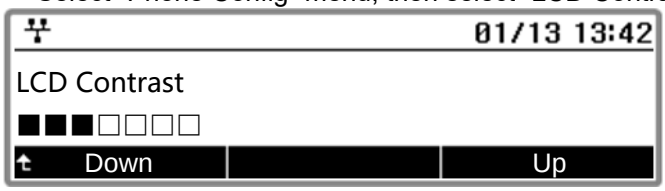
The menu items of the Phone Settings are variable according to the phone model, so the number of menu item may be different by the phone model in the below describes. Please check the menu name when you select a menu item.

4.7.1 LCD Contrast

The contrast of the LCD is adjustable for best viewing angle, and it is supported by 1010i and 1020i.

To adjust the LCD contrast:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “LCD Contrast” menu.



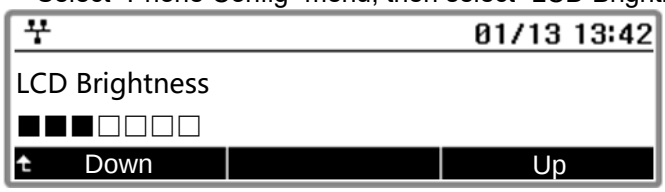
3) Press the Down softkey to decrease the contrast or press the Up softkey to increase the contrast. The VOL ▲▼ button also is used to control the contrast.

4.7.2 LCD Brightness

This menu adjusts the brightness of the LCD, and it is supported by 1030i, 1040i and 1050i.

To adjust the LCD brightness:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “LCD Brightness” menu.



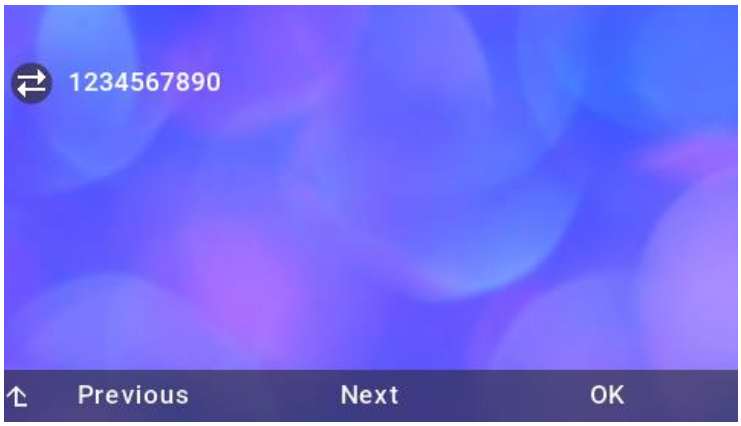
3) Press the Down softkey to decrease the brightness or press the Up softkey to increase the brightness. The VOL ▲▼ button also is used to control the brightness.

4.7.3 Background Image

This menu allows you to change the background picture of the phone. It is supported by 1030i, 1040i and 1050i.

To change the background image:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Background Image” menu.



3) Change the picture using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to set.

4.7.4 Ring Setting

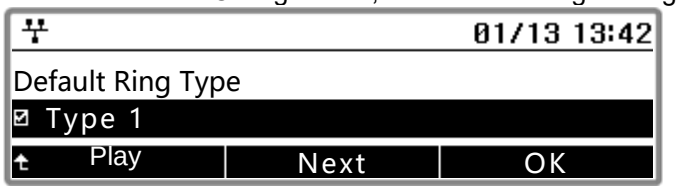
The IP Phone uses one of 8 ring signals (Ring Types) or no ring (“Ringer Off”) for incoming call notification. Each line and Phonebook record may have a different ring type assigned including “Ringer Off” and “Set as Default”. The “Default Ring Type” is used when a ring type is “Set as Default” or not defined for a line or Phonebook record. By assigning different ring signals to closely located phones, lines or Phonebook records, the user can identify the ringing phone, line or caller.

The 1000i supports the 4 types of tone rings and 8 types of wave-form rings. For each line, the ring signal can be immediate or delayed by up to 10 seconds. Delayed ring is defined in one-second increments from 0 to 10 seconds. By assigning different ring signals to closely located phones lines or Phonebook records, the user can identify the ringing phone, line or caller.

4.7.4.1 Default Ring

To select Default Ring Type for the IP Phone:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Ring Setting” menu.



3) Select “Default Ring Type” menu.
4) Select the ring type for the line using Next - Previous softkeys or VOL ▲▼ button. Press the Play softkey to hear the selected tone.
5) Press the OK softkey to change to the selected ring type.

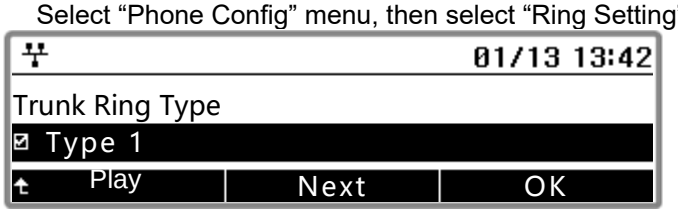
NOTE

- ✓ The Default Ring Type can be set to one of 12 Ring types.
- ✓ The ☑ mark displays in front of the current selection.

4.7.4.2 Trunk Ring Type

To select Trunk Ring Type for the IP Phone:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Ring Setting” menu.



3) Select “Trunk Ring Type” menu.
4) Select the ring type for the line using Next - Previous softkeys or VOL ▲▼ button. Press the Play softkey to hear the selected tone.
5) Press the OK softkey to change to the selected ring type.

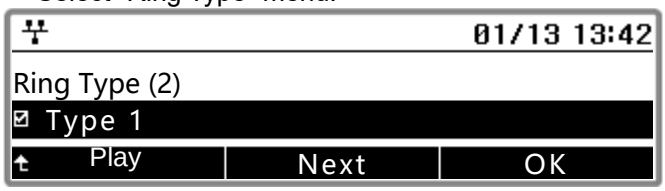
NOTE

- ✓ If no Ring Type or “Set as Default” is assigned for a line, the “Default Ring Type” is used.
- ✓ The Ring Type can be set to one of 12 Ring types, “Set as Default” or “Ringer Off”. When set as Ringer Off, the phone will not ring for incoming calls on the line.
- ✓ The ☑ mark displays in front of the current selection.

4.7.4.3 Line Ring Setting

To adjust Ring Type (tone) for a line :

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Ring Setting” menu.
3) Select “Line *n* Configuration” menu.
4) Select “Ring Type” menu.



5) Select the ring type for the line using Next - Previous softkeys or VOL ▲▼ button. Press the Play softkey to hear the selected tone. The Ringer Off selection turns the ringer off for incoming calls on the line. The “Set as Default” uses the Default Ring Type for incoming calls on the line.
6) Press the OK softkey to change to the selected ring type.

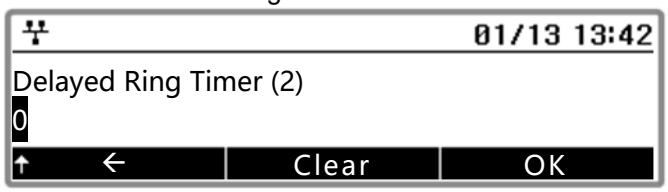
NOTE

- ✓ If no Ring Type or “Set as Default” is assigned for a line, the “Default Ring Type” is used.
- ✓ The Ring Type can be set to one of 12 Ring types, “Set as Default” or “Ringer Off”. When set as Ringer Off, the phone will not ring for incoming calls on the line.
- ✓ The ☑ mark displays in front of the current selection.

4.7.4.4 Delayed Ring

To set Delayed Ring for a line:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Ring Setting” menu.
3) Select “Line *n* Configuration” menu.



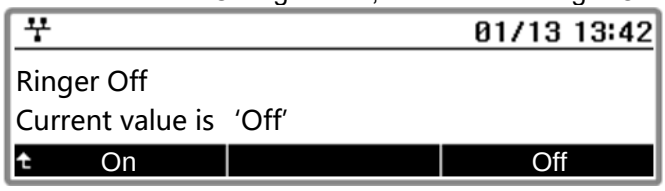
4) Select “Delayed RingTimer” menu.
5) Input the delay time as 0 to 10 seconds in 1-second increaments. A zero (0) provided no delay; the ring signal is immediate.
6) Press the OK softkey to save to the delayed ring setting.

4.7.5 Ringer Off

The IP Phones can disable the ring when a call is incoming.

To set the Ringer Off parameter:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Ringer Off” menu.



3) To enable Ringer Off, press the On softkey or, to disable Ringer Off, press the Off softkey.

4.7.6 Headset Auto Mode

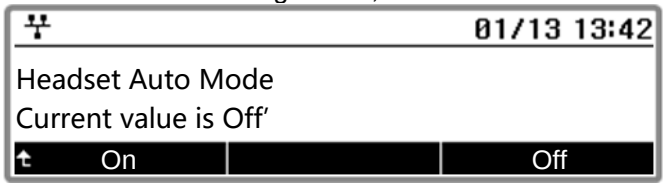
The 1000i series phones support use of a headset.

With Headset Auto Mode enabled, pressing a line button automatically connects audio to the headset.

Also redial, Speed dial, Speed number dialing and click-to-call places calls with audio through the headset.

To enable Headset Auto Mode:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Headset Auto Mode” menu.



3) To enable Headset Auto Mode, press the On softkey or, to disable Headset Auto Mode, press the Off softkey.

4.7.7 Flexible Button Setting

The flexible buttons not assigned as line buttons can access various functions. Table Flexible Button Functions indicates the available functions with a description of each.

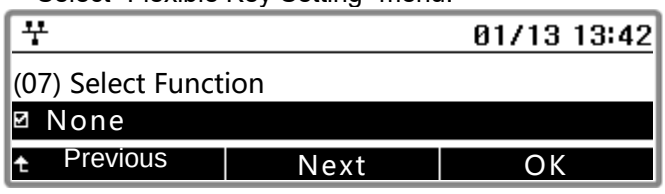
Feature	Description
None	Unassigned
VoIP Line	Assing Voip Line
DSS/BLF	Initiates a call to the DSS/BLF station
Speed Dial	Dials the assigned number
Phonebook	Views the personal phone book
Call Log	Views the call log list
Mute	Toggles the Mute feature on and off
Headset	Toggles the Headset on and off
Redial	Dials the last number dialed by the user.
Call Return	Dials the number of the last answered call
Call Forward	Sets the call forwarding configuration
Conference	1. Initiates a conference call 2. Joins calls in a conference 3. Terminates the conference call
Transfer	Initiates a call transfer
Blind Transfer	Initiates a blind transfer
Hold	Places an active call on hold
Do Not Disturb	Toggles the DND on and off
Ringer Off	Toggles the ringer on and off
Message	Views the number of new and old messages stored in the Voice Mail server.
QueueStatus	Assing the Queue status feature for the status of the logged in Agent
Service	Assing XML service feature
Voice Recording	Records the voice of call to the record server
Adhoc Conerence	Make a conference using the conference server
Feature Key	Assign a feature key funcation for auto dial or led controlled by System
Remote Phonebook	Accesses to the remote phone book.

Table 4.7.7 Flexible Button Functions

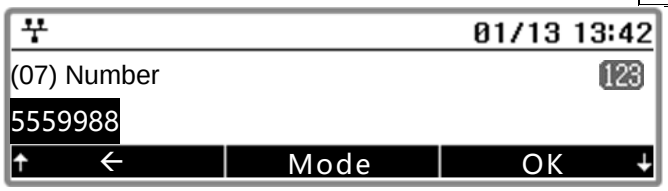
To assign a function to a flexible button:

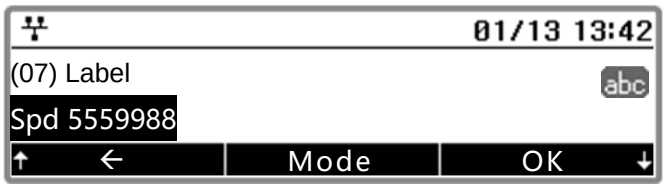
1) Press the Menu button.
2) Select “Phone Config” menu.

3) Select “Flexible Key Setting” menu.



4) And press a flexible button to set a function.
5) Select a function using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey.





6) If the seleted function needs an additional number, then enter the number and press the OK softkey
7) Enter the label and press the OK softkey to save the entry.

NOTE

1010i has 4 flexible buttons, 1020i has 16 flexible buttons, 1030i has 18 flexible buttons, 1040i has 24 flexible buttons and 1050i has 36 flexible buttons.

A 24 button Expansion Module may be connected to the IP Phone expanding the number of Flexible buttons available with the IP Phone by 24.

4.7.8 Time Configuration

Within the Time Configuration menu, the IP Address or FQDN of the desired Simple Network Time Protocol (SNTP) server and the local time zone are defined.

In addition, Daylight Savings Time (DST) is enabled. The following SNTP server addresses or other private SNTP server addresses may be used for Time Configuration:

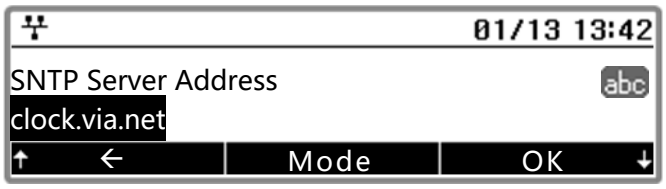
- clock.via.net
- ntp.nasa.gov
- tick.ucla.edu

4.7.8.1 SNTP Server Address

SNTP is a standard protocol that permits a network device to obtain the network's current time. For operation with SNTP, the IP Phones require an SNTP server IP address or FQDN.

To assign the SNTP Server Address:

1) Press the Menu button.
2) Select "Phone Config" menu, then select "Time Configuration" menu.
3) Select "SNTP Server Address" menu.



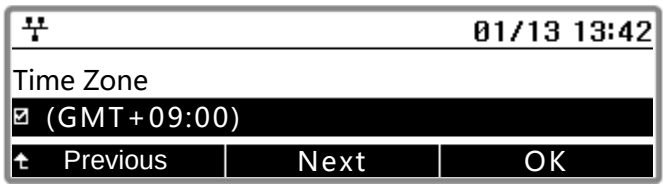
4) Input the SNTP Server Address using the dial pad and press the OK softkey to save your entry.

4.7.8.2 Time Zone

For proper operation of the clock, the IP Phone requires the local time zone.

To enter the Time Zone:

1) Press the Menu button.
2) Select "Phone Config" menu, then select "Time Configuration" menu.
3) Select "Time Zone" menu.



4) Select the time zone desired using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

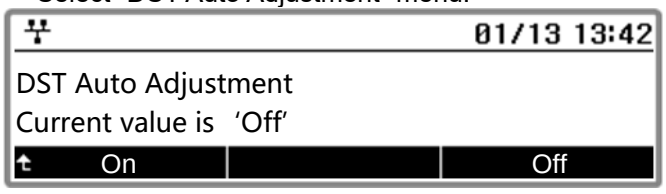
The ☑ mark displays in front of the current selection.

4.7.8.3 DST Auto Adjustment

The IP Phone automatically adjusts for DST when enabled.

To enable/disable automatic day light savings time:

1) Press the Menu button.
2) Select "Phone Config" menu, then select "Time Configuration" menu.
3) Select "DST Auto Adjustment" menu.



4) To enable DST Auto Adjustment, press the On softkey or, to disable DST Auto Adjustment, press the Off softkey.

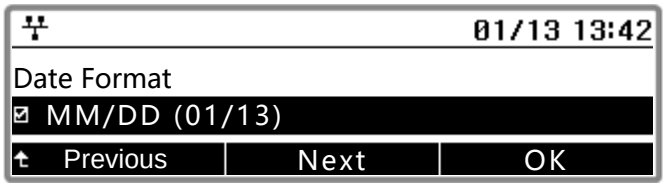
4.7.8.4 Date Format

The format of the LCD date displays in the one of the formats below:

- Disable
- MM/DD (01/13)
- DD/MM (13/01)
- DD m (13 JAN)
- m DD (JAN 13)

To set the date display format:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Time Configuration” menu
3) Select “Date Format” menu.



4) Select the date format using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

The ☑ mark displays in front of the current selection.

4.7.9 S/W Update

Within the S/W Update selection, the type of provisioning (TFTP/HTTPS), the Server Address, HTTPS Verification settings and DHCP Options can be viewed and changed.

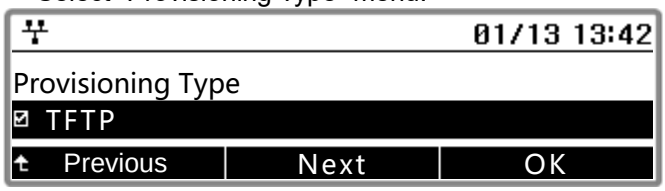
4.7.9.1 Provisioning Type

The 1000i Series phones support two types of provisioning protocols.

- Trivial File Transfer Protocol (TFTP) – a simple transfer mechanism using UDP but lacks security and TFTP ports are often blocked at firewalls.
- Hyper-Text Transfer Protocol/Secure (HTTP/S) – employs TCP and using SSL (Secure Sockets Layer) encryption is a more secure method. In addition, ports employed by HTTPS are commonly available.

To assign the provisioining protocol:

1) Press the Menu button and select “Phone Config” menu, then select “S/W Update” menu.
2) Select “Provisioning Type” menu.



3) Select the Provisioning Type (TFTP or HTTP/S) using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

Changing the Provisioning Type requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

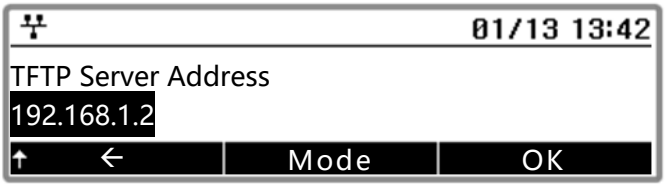
The ☑ mark displays in front of the current selection.

4.7.9.2 TFTP Server Address

With TFTP provisioning selected, the IP address or FQDN of the TFTP server must be defined. The TFTP Server Address may be retrieved automatically from the DHCP server or set manually.

To enter a TFTP Server Address:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “TFTP Server Address” menu.



4) Input the TFTP Server Address using the dial pad and press the OK softkey to save your entry.

NOTE

Changing the TFTP address requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

4.7.9.3 HTTPS Provisioning

The HTTP/HTTPS server URL is a required parameter for HTTP/HTTPS provisioning. Normally, HTTP employs TCP port 80, and HTTPS employ port 443. In addition, if required by the HTTP/HTTPS server for authentication, a Username and Password are assigned.

NOTE

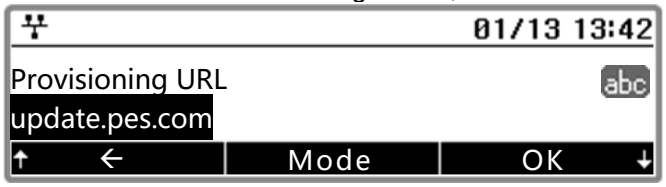
When using HTTPS, the server must support SSL (Secure Sockets Layer) for proper operation.

Provisioning URL

The Provisioning URL is the URL of the HTTP/HTTPS server.

To assign the provisioning server URL:

1) Press the Menu button and select “Phone Config” menu, then select “S/W Update” menu.
2) Select “HTTPS Provisioning” menu, then select “Provisioning URL” menu.



3) Input the provisioning server URL using the dial pad and press the OK softkey to save your entry.

NOTE

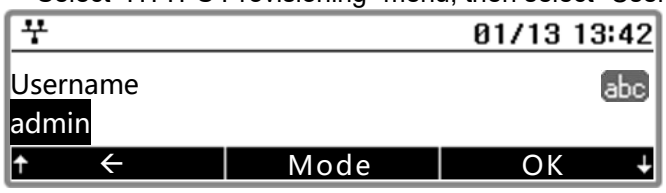
Changing the provisioning server URL requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

Username

The HTTP/HTTPS server requires a Username (Id) and Password for authentication prior to allowing file access.

To set the Username:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “HTTPS Provisioning” menu, then select “Username” menu.



4) Input the username using the dial pad and press the OK softkey to save your entry.

NOTE

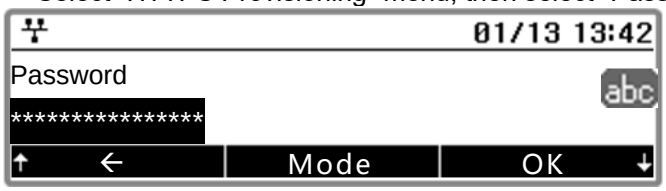
Changing the username requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

Password

The HTTP/HTTPS server requires a Username (Id) and Password for authentication prior to allowing file access.

To set the Password:

1) Press the Menu button and select “Phone Config” menu, then select “S/W Update” menu.
2) Select “HTTPS Provisioning” menu, then select “Password” menu.



3) Input the password using the dial pad and press the OK softkey to save your entry.

NOTE

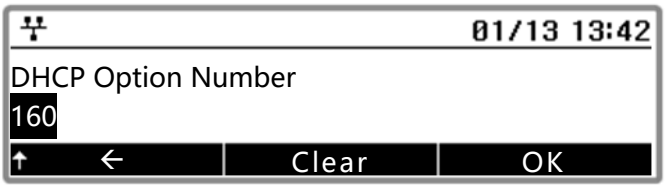
Changing the password requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

HTTPS URL DHCP Option Number

An unused DHCP option can be used to define the HTTPS provisioning server URL. As a default, the IP Phone uses DHCP option 160.

To modify the DHCP option used for the HTTPS URL:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “HTTPS Provisioning” menu, then select “DHCP Option Number” menu.



4) Input the DHCP option, 1 to 255 using the dial pad and press the OK softkey to save your entry.

NOTE

Changing the DHCP option requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

Checking Certificates

1000i IP Phones can verify specific characteristics of the HTTPS server and the Certificate of Authority stored on the server.

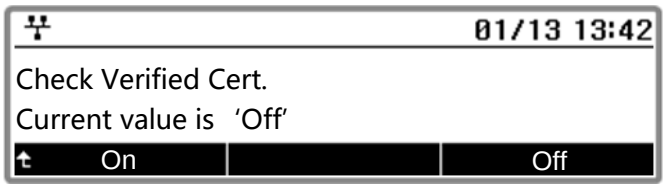
Verifiable characteristics are:

- Valid Certificate signature and Certificate of Authority expiration date
- HTTPS server name

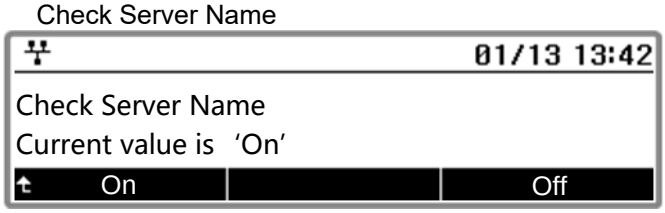
If the verification fails, IP Phones terminate the HTTPS session.

To modify the verification options:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “HTTPS Provisioning” menu, then select “Checking Certificates” menu.
4) Select the desired option using the 1 or 2 button, or the Next - Previous softkeys or VOL ▲▼ button
5) press the OK softkey.
6) Check Verified Cert



7) Check Server Name



For each option, press the softkey to enable the check, or press the Off softkey to disable the check.

NOTE

Changing the verification options requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

The display will indicate the present value (On or Off) for each option.

4.7.9.4 S/W Download Flag

When Power is applied or a reboot is processed, the IP Phone automatically checks and downloads IP Phone Software from the defined provisioning server.

The IP Phone will download software if the software versions are different and the S/W Download flag is enabled. Note this menu item displays only when the Download Flag function is enabled in the Web Manager.

To set the S/W Download Flag:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “S/W Download Flag” menu.

4) To enable the S/W download, press the On softkey or, to disable the S/W download, press the Off softkey.

NOTE

Changing the S/W download requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice.

The IP phone must reboot to utilize the new values entered.

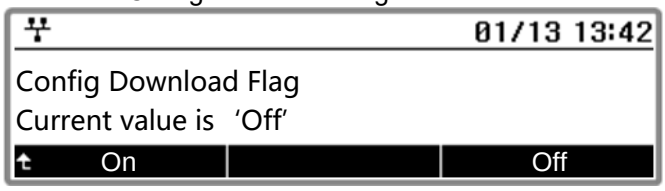
4.7.9.5 Config Download Flag

When Power is applied or a reboot is processed, the IP Phone automatically downloads a Global (system) and a Phone specific (MAC) Configuration file from the defined provisioning server.

To download the configuration files during provisioning, the Config Download flag must be enabled. Note this menu item displays only if the Download Flag function is enabled using the Web Manager.

To set the Config Download Flag:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “Config Download Flag” menu.



4) To enable the Config download, press the On softkey or, to disable the Config download, press the Off softkey.

NOTE

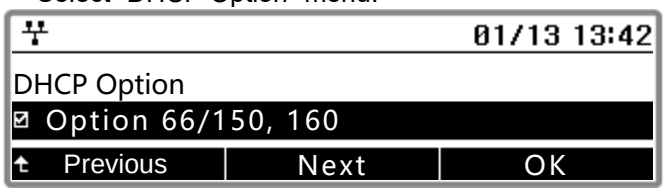
Changing the config download requires a reboot of the IP phone; when you leave the menu, you will receive a reboot notice. The IP phone must reboot to utilize the new values entered.

4.7.9.6 DHCP Option

The IP Phone supports DHCP Options 66 & 150 (TFTP server) and a custom Option, default 160, for the HTTP/HTTPS URL to define the provisioning server. In some situations, it is useful to disable use of the DHCP Options.

To set the DHCP Options flag:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu.
3) Select “DHCP Option” menu.



4) Select the DHCP option desired using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry. Options available include “Don’t use any Option”, “Option 66/150 only”, “Option 160 only”, and “Option 66/150, 160”.

NOTE

Changing the DHCP Options requires a reboot of the IP Phone; when you leave the menu, you will receive a reboot notice. The IP Phone must reboot to utilize the new values entered.

The ☑ mark displays in front of the current selection.

4.7.9.7 Update Now

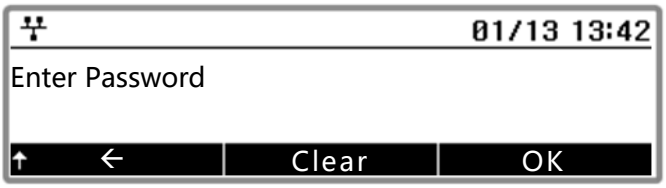
Without the reboot, the IP Phone checks whethter there is the configuration and the latest S/W or not.

To check the configuration and S/W

1) Press the Menu button.
2) Select “Phone Config” menu, then select “S/W Update” menu
3) Select “Updaet Now” menu.

4.7.10 Lock Outgoing Call

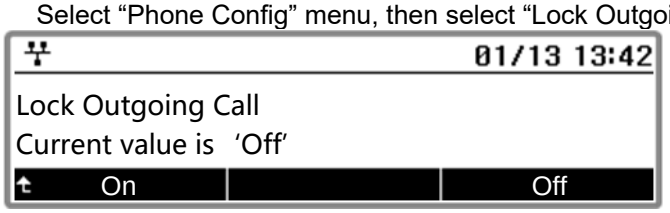
The IP Phone can be configured to allow outgoing calls only if a password is entered. When enabled, the LCD screen will display the password request as shown and the password must be entered prior to connecting for outgoing call.



To activate this feature, the LCD password must be set, and Lock Outgoing Call must be enabled.

To enable/disable the Lock Outgoing Call feature:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Lock Outgoing Call” menu.



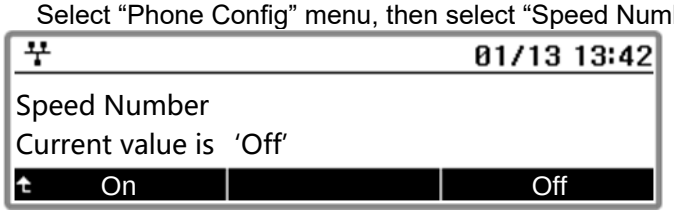
3) To enable the Lock Outgoing Call, press the On softkey or, to disable the Lock Outgoing Call, press the Off softkey.

4.7.11 Speed Number

1000i series phone includes a Phonebook that supports up to 1000 entries. Each Phonebook record includes a three (3) digit Speed number. When enabled, dialing the Speed Number (000 ~ 999) sends the contact information from the associated Phonebook record.

To enable/disable Speed Number operation:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Speed Number” menu.



3) To enable the speed number, press the On softkey or, to disable the speed number, press the Off softkey.

NOTE

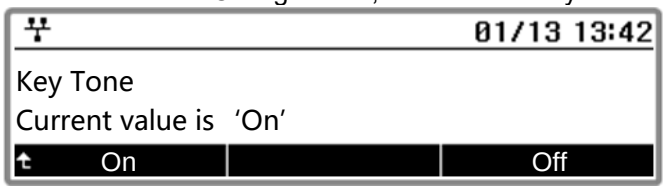
When enabled, dialing an empty Speed Number terminates the call and the empty status of the Speed Number displays for approximately 2 seconds.

4.7.12 Key Tone

Normally, when a button on the phone is pressed a tone is generated to confirm to the user that the phone has recognized the key depression. If desired, this tone can be disabled.

To enable/disable the Key Tone:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Key Tone” menu.



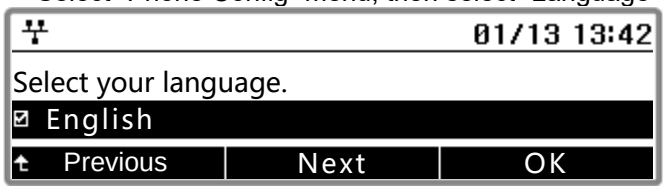
3) To enable the key tone, press the On softkey or, to disable the key tone, press the Off softkey.

4.7.13 Language

The IP Phone firmware includes an xml file, which, among other things, determines the languages available to the phone. When the xml file contains more than one (1) language, the Language parameter will be available.

To select an available language:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Language” menu.



3) Select a language from the list using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

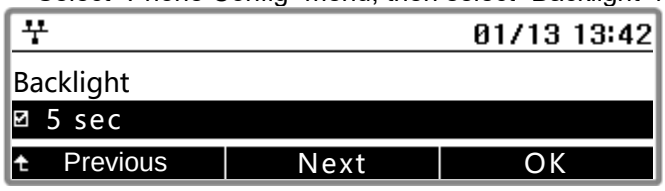
The ☑ mark displays in front of the current selection.

4.7.14 Backlight

In idle screen, the 1000i series lights up the LCD for the configured time. If ‘Off’ is selected, IP Phone doesn’t light up the LCD during any operation.

To configure the backlight on time:

1) Press the Menu button.
2) Select “Phone Config” menu, then select “Backlight” menu.



3) Select a time to turn on the backlight of LCD from the list using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

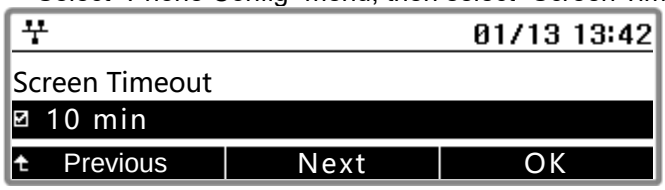
The ☑ mark displays in front of the current selection.

4.7.15 Screen Timeout

1030i, 1040i and 1050i turn off the LCD to save the LCD lifetime when it doesn't work anything for a long time.

To configure the screen timeout:

1) Press the Menu button.
2) Select "Phone Config" menu, then select "Screen Timeout" menu.



3) Select a time to turn off the LCD from the list using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry.

NOTE

The ☑ mark displays in front of the current selection.

4.8 Call Preferences

Call Preferences define various call handling features of the IP Phones.

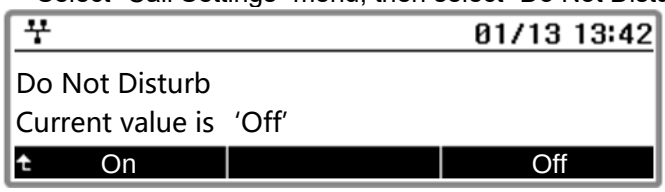
4.8.1 Do Not Disturb

When the DND (Do Not Disturb) feature is active, the IP Phone does not notify the user of an incoming call. The IP Phone does not ring, and the associated line LED does not flash.

This allows you to work without interruption from an incoming call. The call server configuration determines routing of the incoming call while your phone is in DND. To activate DND, DND must be enabled.

To enable the DND operation:

1) Press the Menu button.
2) Select "Call Settings" menu, then select "Do Not Disturb" menu.



3) To enable the DND, press the On softkey or, to disable the DND, press the Off softkey.

4.8.2 Call Waiting

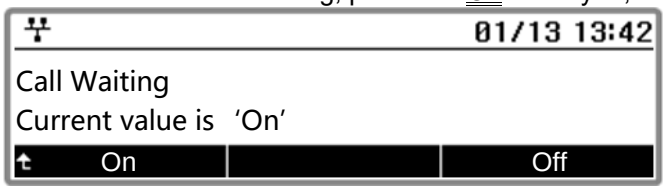
With Call Waiting configured, while on a call the IP Phone alerts the user of an incoming call. The line button flashes to indicate the new call, and an optional beep tone is delivered over the existing conversation.

In addition, the LCD displays the incoming caller id during the ring signal, so you know who is calling before you interrupt the current call.

If Call Waiting is disabled, the IP Phone will reject the new call. The call server configuration determines routing of the incoming call.

To enable/disable Call Waiting:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Call Waiting” menu.
3) To enable the call waiting, press the On softkey or, to disable the call waiting, press the Off softkey.



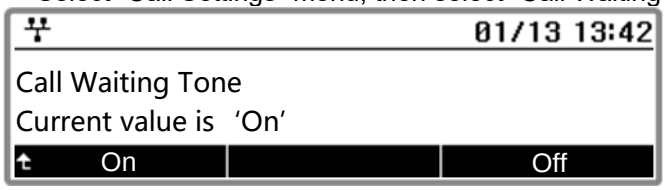
4.8.3 Call Waiting Tone

While off-hook, the IP Phone can provide a tone over the existing conversation to indicate a call waiting.

To receive the tone, Call Waiting must also be enabled.

To enable/disable Call Waiting Tone:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Call Waiting Tone” menu.



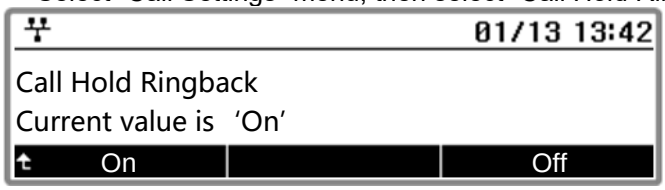
3) To enable the call waiting tone, press the On softkey or, to disable the call waiting tone, press the Off softkey.

4.8.4 Call Hold Ringback

With Call Hold Ringback enabled, the IP Phone rings as a reminder of calls on hold when the user returns the IP Phone to idle.

To enable/disable Call Hold Ringback:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Call Hold Ringback” menu.



3) To enable the call hold ringback, press the On softkey or, to disable the call hold ringback, press the Off softkey.

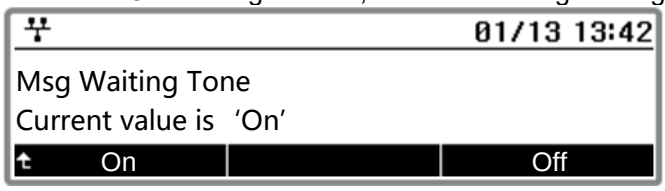
4.8.5 Message Waiting Tone

When notification of a new message is received, the Message LED flashes. As an option, a ‘stutter’ dial tone provides an audible indication for a message waiting.

When the user goes off-hook to place a call, the user receives stutter dial tone.

To enable/disable stutter dial tone for a message wait indication:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Msg Waiting Tone” menu.



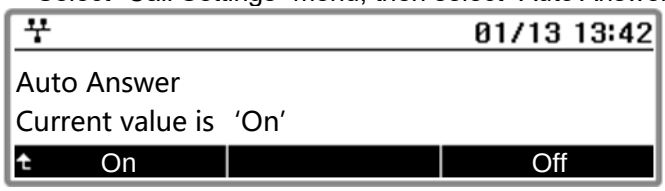
3) To enable the stutter dial tone, press the On softkey or, to disable the stutter dial tone, press the Off softkey.

4.8.6 Auto Answer

When idle, the IP Phone can connect an incoming caller to the speakerphone or headset automatically.

To enable/disable Auto Answer:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Auto Answer” menu.



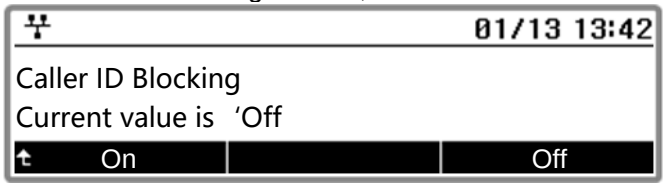
3) To enable the auto answer, press the On softkey or, to disable the auto answer, press the Off softkey.

4.8.7 Caller ID Blocking

The IP Phone normally sends the assigned display name as the SIP caller id for outgoing calls. Optionally, the IP Phone can send “anonymous” as the caller id in SIP message headers.

To enable/disable blocking of caller id:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Caller ID Blocking” menu.



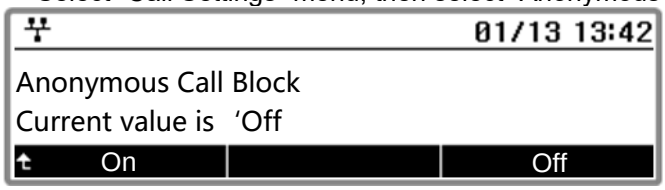
3) To enable the caller ID blocking, press the On softkey or, to disable the caller ID blocking, press the Off softkey.

4.8.8 Anonymous Call Block

The 1000i phones can block incoming calls with an anonymous caller id.

To enable/disable Anonymous Call Blocking:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Anonymous Call Block” menu.



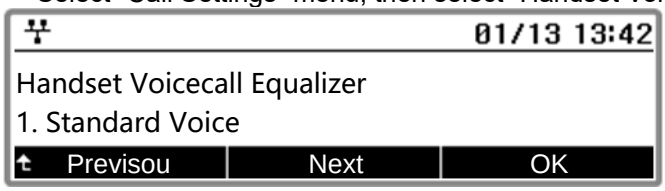
3) To enable the anonymous call blocking, press the On softkey or, to disable the anonymous call blocking, press the Off softkey.

4.8.9 Handset Voicecall Equalizer

The 1000i phones can set the handset voice call equalizer.

To configure handset voice call the equalizer:

1) Press the Menu button.
2) Select “Call Settings” menu, then select “Handset Voicecall Equalizer” menu.



3) Select the desired value of equalizer using Next - Previous softkeys or VOL ▲▼ button and press the OK softkey to save the entry. Options available include “Standard Voice”, “Sharp voice” and “Soft Voice”.

4.9 Phonebook

The IP Phone supports two Phonebooks, an internal Phonebook and a Remote (external Phonebook. The internal Phonebook is stored and managed at the IP Phone. The Remote Phonebook is managed externally, and the IP Phone can access the Phonebook as an xml file to search and call contacts in the Remote Phonebook.

4.9.1 Internal Phonebook

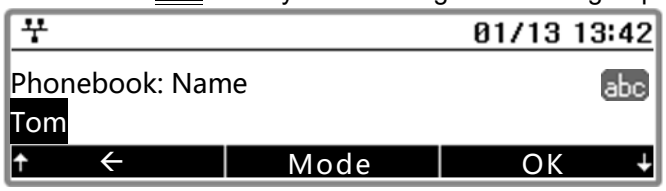
IP Phones include an internal Phonebook of names and numbers, or SIP user ids for up to 1000 contacts. The Phonebook menu is accessed through the LCD Menu, with the Down navigation button or, when assigned, a Phonebook flexible button.

In the Phonebook menu, records may be added, edited, deleted or searched. The phone number or SIP user id in the Phonebook is the contact information used with Speed Number dialing. Each contact can have a Ring type assigned. When a call is received from the contact, the specified ring signal is employed.

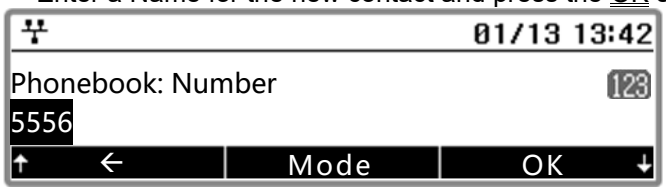
4.9.1.1 Adding a Phonebook record

To add a new contact:

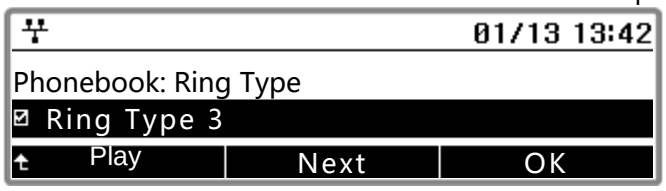
1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Press the Add softkey after moving to the next group of softkeys using the Right navigation button.



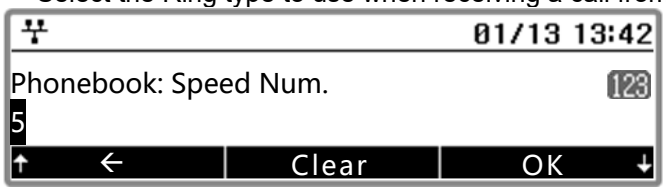
4) Enter a Name for the new contact and press the OK softkey.



5) Enter a Number or SIP user ID for the contact and press the OK softkey.



6) Select the Ring type to use when receiving a call from this contact.

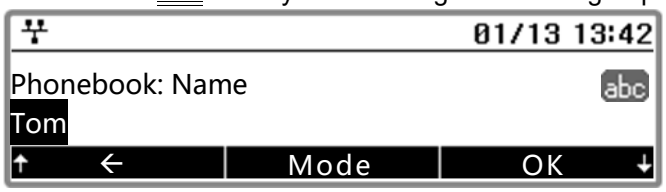


7) Enter a Speed Number (000 ~ 999) for the new contact and press the OK softkey.
8) The IP phone adds the new contact to the Phonebook and returns to the Phonebook list.

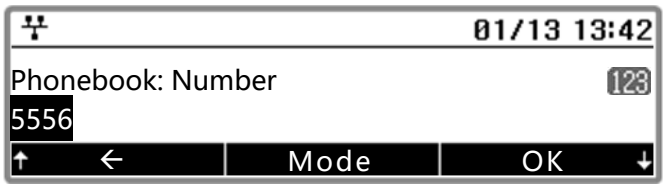
4.9.1.2 Editing a record

To edit a Phonebook record:

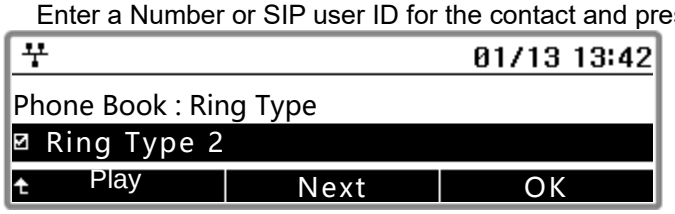
1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Press the Edit softkey after moving to the next group of softkeys using the Right navigation button.



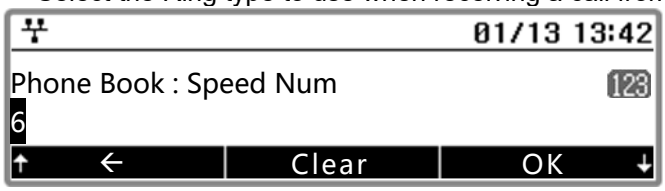
4) Enter a Name for the new contact and press the OK softkey.



5) Enter a Number or SIP user ID for the contact and press the OK softkey.



6) Select the Ring type to use when receiving a call from this contact.



7) Enter a Speed Number (000 ~ 999) for the new contact and press the OK softkey.
8) The IP phone adds the new contact to the Phonebook and returns to the Phonebook list.

NOTE

If no ring type is defined for a phonebook entry, the default ring type is used.

4.9.1.3 Deleting a record

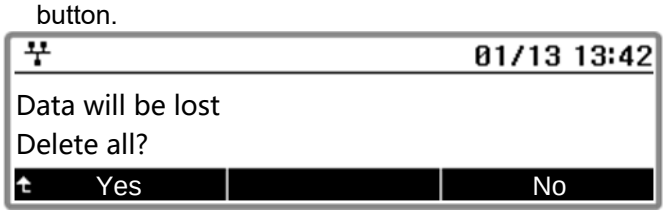
To delete a Phonebook record:

1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Select a contact to delete using the Next - Previous softkeys or VOL ▲▼ button.
4) Press the Delete softkey and confirm by selecting the Yes softkey. Caution this action deletes the record from the Phonebook.
5) The IP phone deletes the contact from the Phonebook and returns to the Phonebook list.

4.9.1.4 Deleting all records

To delete all Phonebook records:

1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Press the Del.All softkey after moving to the next group of softkey using the Right navigation button.

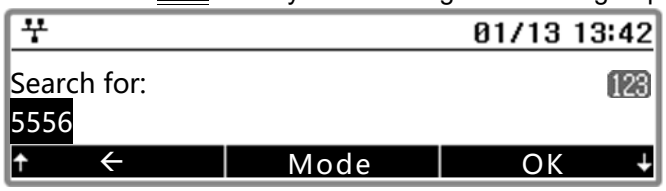


4) Press the Yes softkey to confirm deletion of all contacts.
5) The IP phone deletes all contacts from the Phonebook and returns to the Phonebook list.

4.9.1.5 Searching the Phonebook

To search for the Phonebook using either Name or Number:

1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Press the Find softkey after moving to the next group of softkey using the Right navigation button.



4) Input a search keyworkd and press the OK softkey.
5) The search result displays showing the speed number name and number.

4.9.1.6 Dialing from the Phonebook

To dial a number from the Phonebook:

1) Press the Menu button.
2) Select “Directory” menu, then select “Phonebook” menu.
3) Select a contact to dial using the Next - Previous softkeys or VOL ▲▼ button and press the Dial softkey to place the call.

4.9.2 Remote Phonebook

The Remote Phonebook is managed external to the IP Phone. The IP Phone accesses the Remote Phonebook with xml file to search and call contacts from the Phonebook. This menu is displayed on LCD when Remote Phonebook settings have been configured in the Web Manager.

To access the Remote Phonebook:

1) Press the Menu button.
2) Select “Directory” menu, then select “Remote Phonebook” menu.
3) Enter a searched word and press the OK softkey. The searched phonebook items will be shown.

NOTE

This feature needs a remote phone book server.

4.10 Call Log

The IP Phone supports the call log.

To show the call log:

1) Press the Menu button.
2) Select “Directory” menu, then select “Call Log” menu.

To make a call from the call log:

1) Press the Dial softkey to make a call to the selected phone number.

To change the category:

1) Press the Category softkey to change the category of the call log. The category is changed in order “All Calls”, “Missed Calls”, “Received Calls” and “Placed Calls”

To save a selected call log to the internal phonebook:

1) Press the Save softkey.
2) Enter a name and press the OK softkey.
3) Enter a number and press the OK softkey.
4) Select a ring type and press the OK softkey.
5) Enter a speed number and press the OK softkey.

To edit a call log:

1) Press the Edit softkey.
2) Edit the number
3) Press the Dial softkey to make a call to the edited number.
4) Press the Save softkey to save the edited number to the internal phonebook.

To delete a selected call log:

1) Press the Delete softkey.
2) Press the Yes softkey to confirm.

To delete all call log:

1) Press the Del.All softkey.
2) Press the Yes softkey to confirm.

4.11 Status

This menu group includes information, default setting, lock/unlock configuration, and rebooting.

4.11.1 Information

The IP Phone shows the informations of the phone.

To show the information:

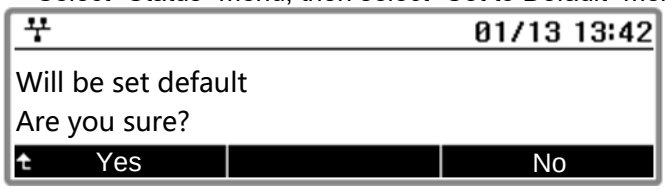
1) Press the Menu button.
2) Select “Status” menu, then select “Information” menu.
3) Press the Next - Previous softkeys or VOL ▲▼ button to change the shown informations.

4.11.2 Set to Default

The IP Phone configuration can be returned to the original factory defaults.

To return to the default configuration:

1) Press the Menu button.
2) Select “Status” menu, then select “Set to Default” menu.



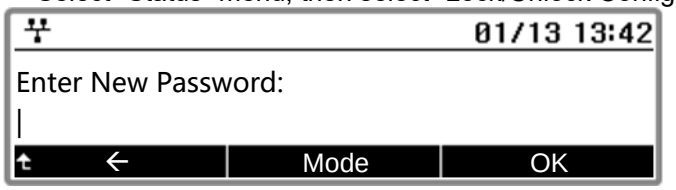
3) Press the Yes softkey to set the IP phone to the default configuration or press the No softkey to cancel.

4.11.3 Lock/Unlock Config

Some important menu items should be restricted from any access by someone. To do this, a passcode needs to be set. And to access a restricted menu items, it should be unlocked by entering a valid passcode. If the phone exists from the menu, it is locked again automatically.

To set a new passcode:

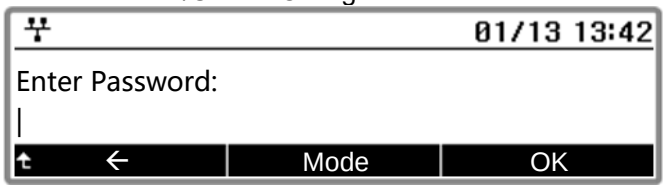
1) Press the Menu button.
2) Select “Status” menu, then select “Lock/Unlock Config” menu.



3) Enter a passcode and press the OK softkey.
4) Enter the same passcode one more time and press the OK softkey.
5) If you failed, please try again with a valid passcode.

To unlock the phone:

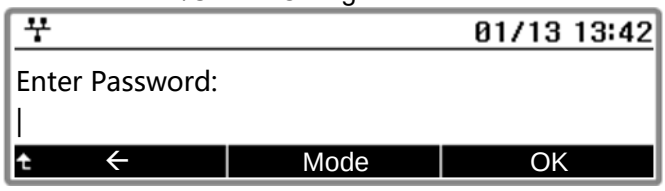
1) Press the Menu button.
2) Select “Status” menu.
3) Select “Lock/Unlock Config” menu.



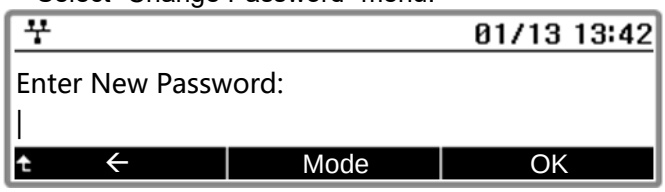
4) Enter the passcode and press the OK softkey.
5) If you enter a valid passcode, the phone will display the “Config is unlocked” message. If you see any error message, please try again with a valid passcode.

To change the passcode:

1) Press the Menu button.
2) Select “Status” menu.
3) Select “Lock/Unlock Config” menu.



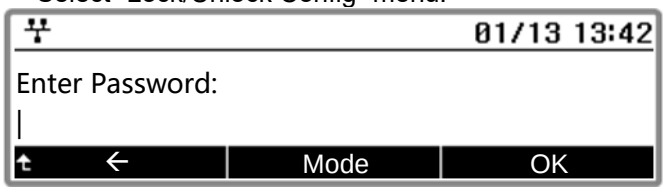
4) Enter the passcode and press the OK softkey.
5) If you enter a valid passcode, the phone will display the “Config is unlocked” message. If you see any error message, please try again with a valid passcode.
6) Select “Status” menu.
7) Select “Lock/Unlock Config” menu.
8) Select “Change Password” menu.



9) Enter a passcode and press the OK softkey.
10) Enter the same passcode one more time and press the OK softkey.
11) If you enter a valid passcode, the phone will display the “Password changed” message. If you see any error message, please try again with a valid passcode.

To remove the passcode:

1) Press the Menu button.
2) Select “Status” menu.
3) Select “Lock/Unlock Config” menu.



4) Enter the passcode and press the OK softkey.
5) If you enter a valid passcode, the phone will display the “Config is unlocked” message. If you see any error message, please try again with a valid passcode.
6) Select “Status” menu.
7) Select “Lock/Unlock Config” menu.
8) Select “Remove Password” menu.

The phone will display the “Password is disabled” message.

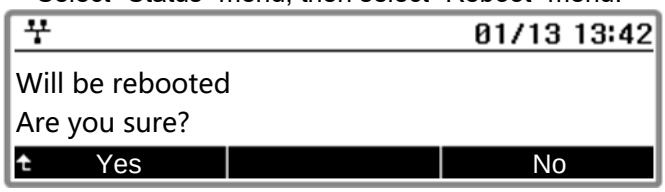
4.11.4 Reboot

After changing certain parameters, the IP Phone will require a reboot to utilize the settings. These settings include the following:

- Network Configuration settings
- Call Server Address settings
- Software Update settings
- Default settings

To reboot the IP Phone:

1) Press the Menu button.
2) Select “Status” menu, then select “Reboot” menu.



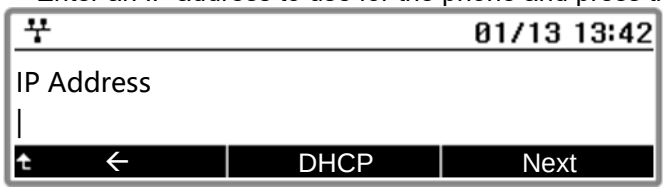
3) Press the Yes softkey to reboot the IP phone, or press the No softkey to cancel.
4) If you press the No softkey in the above step, move to another parameter with the navigation buttons and continue the session or press the Menu button to exit the configuration session without activating the changes. Any parameter changes will update during the next reboot.

4.11.5 Easy set up

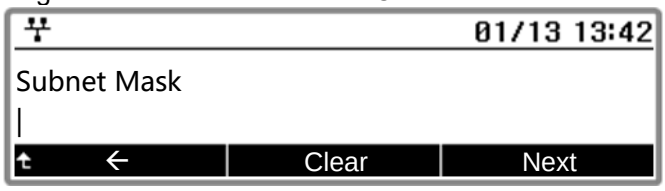
To set the phone easily, you can set only the mandatory configurations first, and other configutaions can be get from a provisioning server. If you use this menu, your phone will be set to default.

To set your phone using the easy set up:

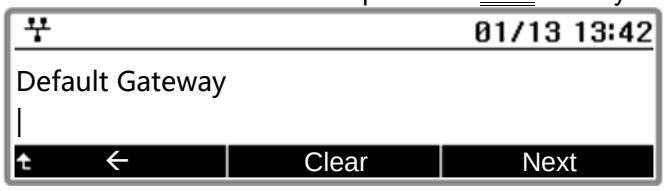
1) Press the Menu button and select “Status” menu, then select “Easy Setup” menu.
2) Enter an IP address to use for the phone and press the Next softkey, or press the DHCP softkey to



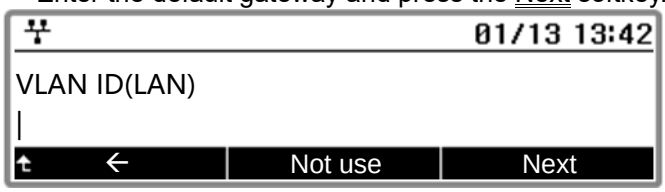
3) get an IP address from a DHCP server.

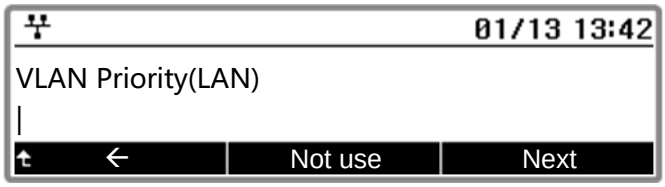


4) Enter the subnet mask and press the Next softkey.

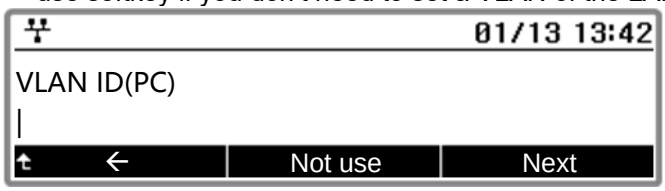


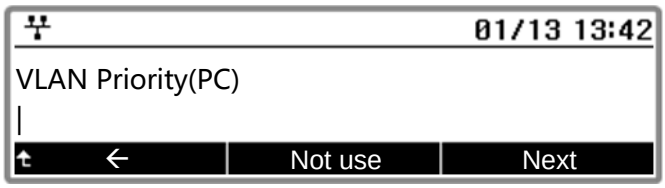
5) Enter the default gateway and press the Next softkey.



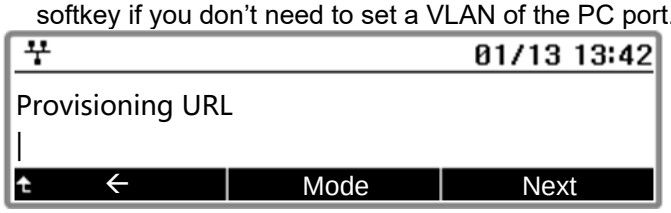


6) Enter the VLAN ID & VLAN Priority for the LAN port and press the Next softkey or press the Not use softkey if you don't need to set a VLAN of the LAN port.

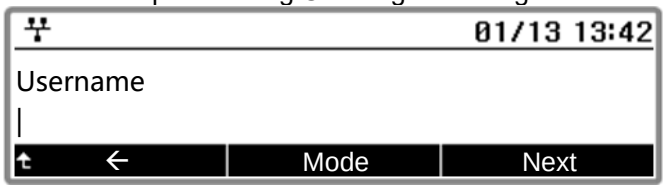


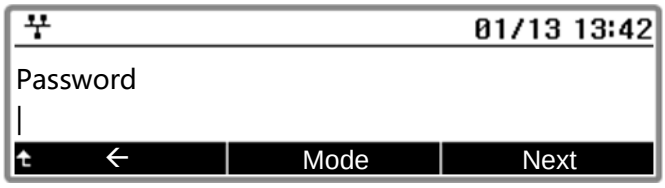


7) Enter the VLAN ID & VLAN Priority for the PC port and press the Next softkey or press the Not use softkey if you don't need to set a VLAN of the PC port.

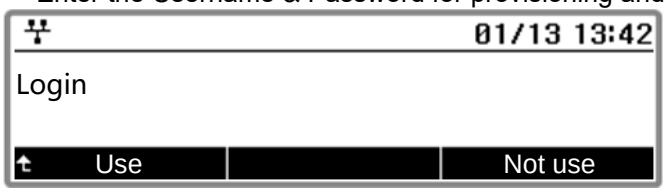


8) Enter the provisioning URL to get a configuration file and press the Next softkey.





9) Enter the Username & Password for provisioning and press the Next softkey.



10) To use the hot-desking feature, press the Use softkey. Or not, press the Not use softkey.
11) To use the web server, press the Use softkey. Or not, press the Not use softkey.
12) Reboots the phone by pressing the OK softkey.

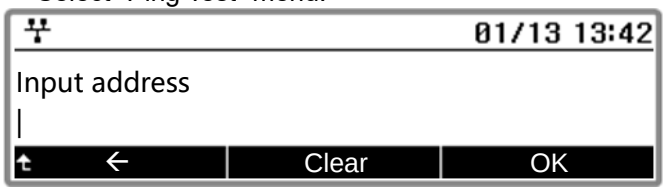
Please contact your administrator if you have any problem.

4.11.6 Diagnosis

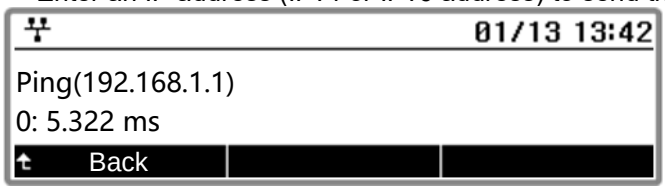
To check the phone, some diagnosis tools are provieded.

To ping a destination IP address:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Ping Test” menu.



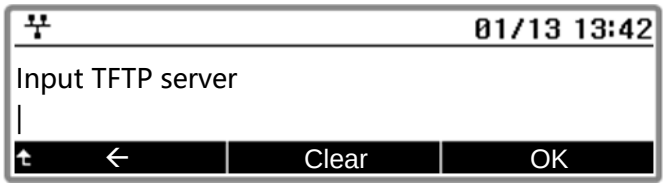
4) Enter an IP address (IPv4 or IPv6 address) to send the ping request and press the OK softkey.



5) The result is shown.

To dump a log:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Dump Log” menu.
4) Wait for a while.



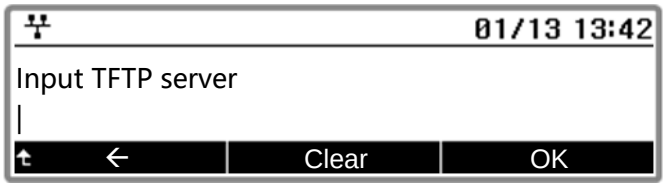
5) Enter an TFTP server address (IPv4 or IPv6 address) to send the log and press the OK softkey.
6) Check the downloaded log file from your TFTp server.

To start capture the packets from/to the phone:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Packet Capture” menu, then select “Start” menu.
4) The phone will start the capturing of packets.

To stop capture the packets and download:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Packet Capture” menu, then select “Stop” menu.
4) Select “Download” menu.



5) Enter an TFTP server address (IPv4 or IPv6 address) to send the log, and press the OK softkey.
6) Check the downloaded log file from your TFTP server.
7) Select “Delete” menu to remove the temporary log file saved in the phone.

To start port mirroring from the LAN port to the PC port:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Port Mirroring” menu, then select “Start” menu.
4) The PC connected to the PC port of the phone can capture the packets from/to the phone.

To stop the port mirroing:

1) Press the Menu button.
2) Select “Status” menu, then select “Diagnosis” menu.
3) Select “Port Mirroring” menu, then select “Stop” menu.

5 Using Web Manager for Configuration

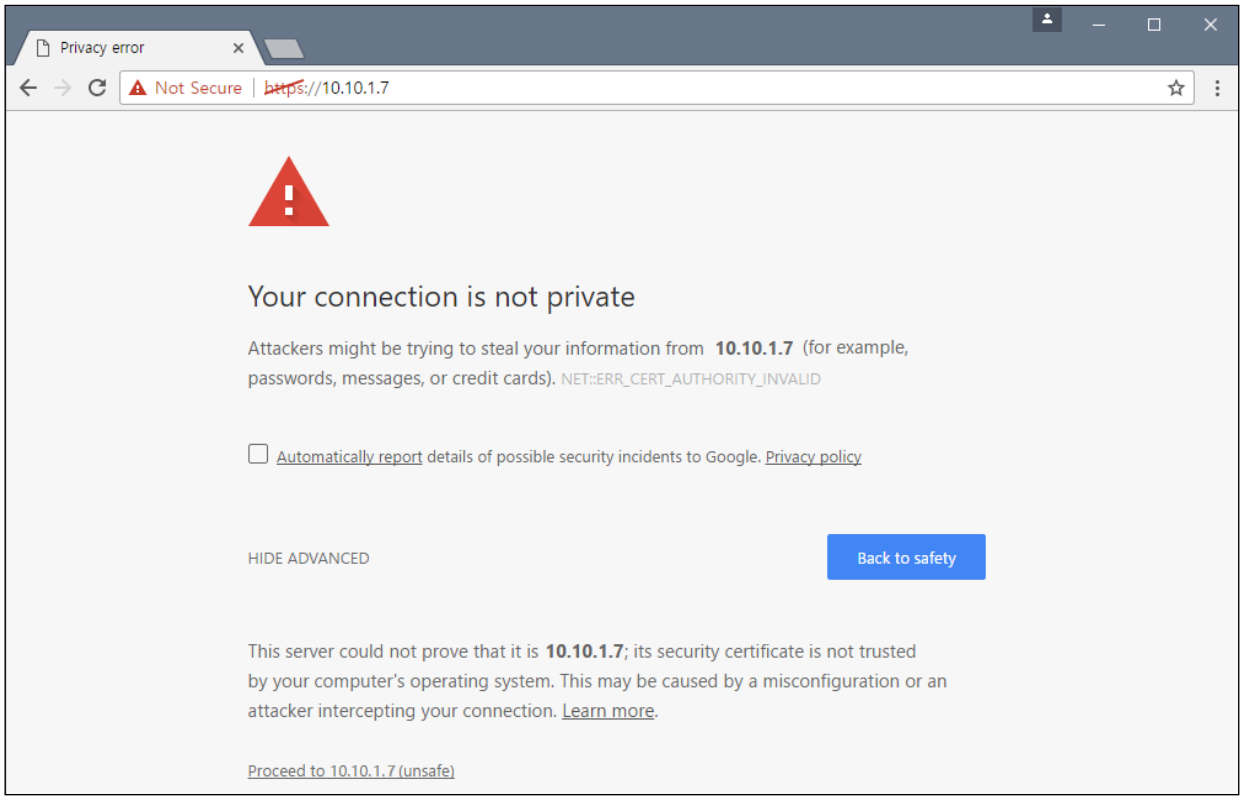
5.1 Login

The Web Manager is a web-based tool for configuring the IP Phone parameters. The Web Manager can access parameters not available through the LCD Configuration Menu. Appendix C. Using the XML configuration file contains information about the parameters to set. In addition to remote access to the IP Phone configuration, Web Manager is a convenient tool when configuring some parameters of multiple IP Phones or a single IP Phone.

For the 1000i series, the default of web setting is 'Disable'. To access the web, change to Menu> Network> 12. WEB: Disable > HTTPS on the terminal.

To access Web manager:

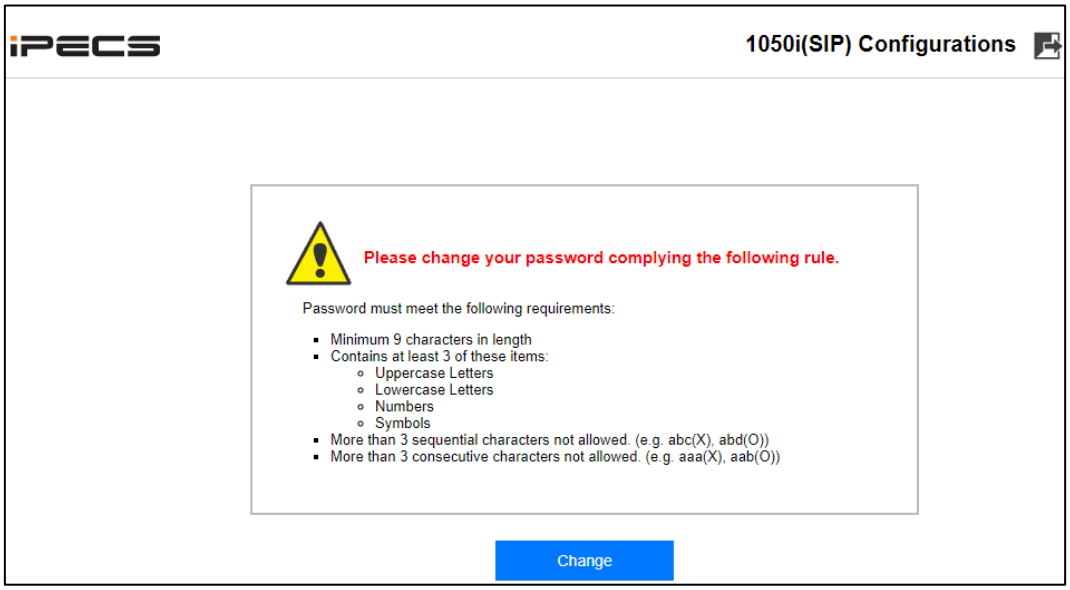
1) Access the IP Phone's IP address with a web browser. (ex, https://10.10.1.7)
   * In order to access the web manager operating in IPv6, you need to input an IPv6 address into the browser. (ex, https://[2001:1:1:a123:b123:c123:d123:000a])
2) Proceed by pressing "Proceed to … (unsafe)" if no certificate is installed.
3) Enter valid user name and password. If you try to access with the default account like this, you must change the password in the next step.
4) Enter a username and new password once you change the default password.



5) Click "Continue to this website (not recommended)", if the login page shows 'Privacy error' as above

To change the default password:

1) When you try to access with the default account like “user/lip”, you should change your password in the next step.



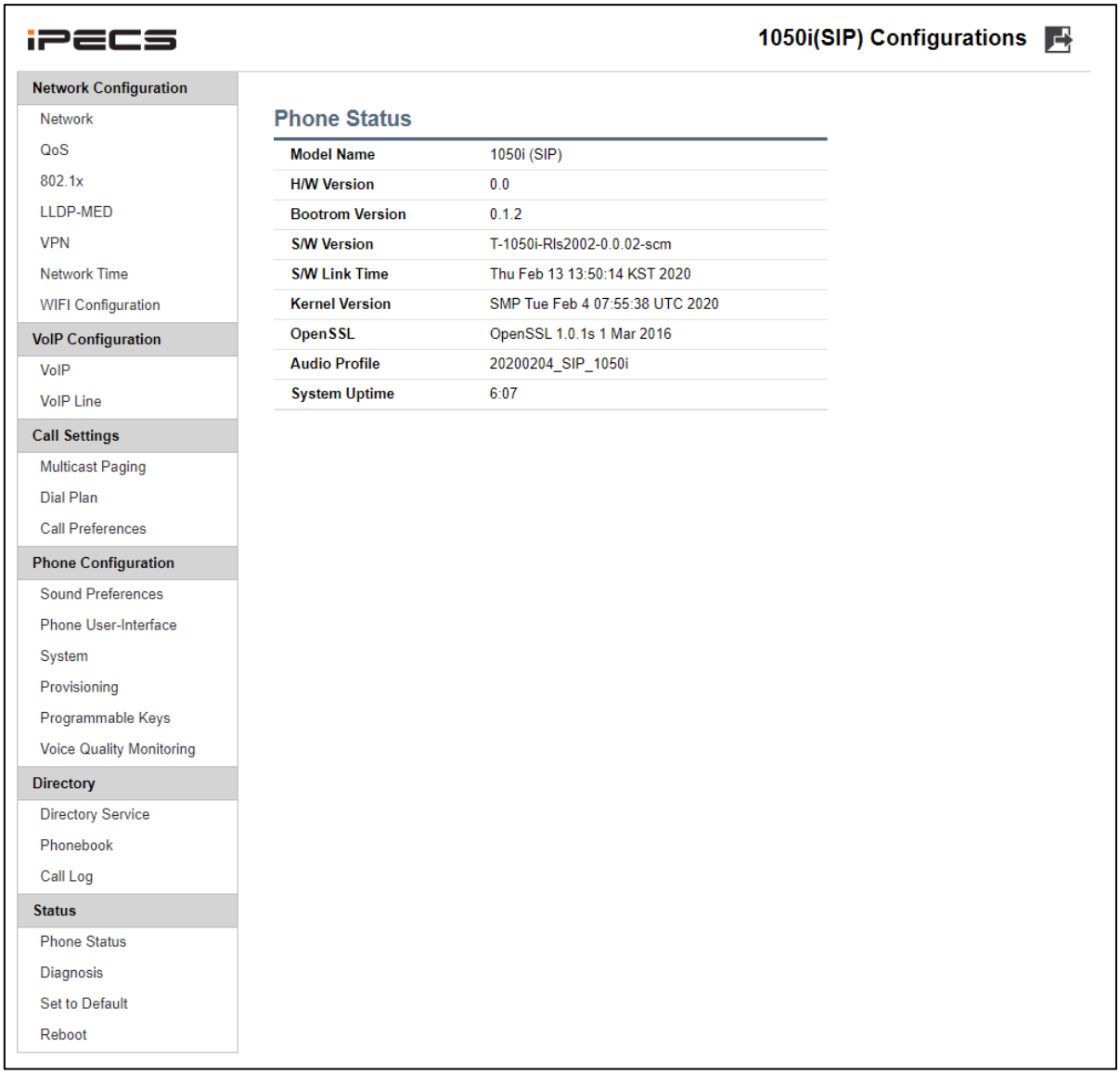
2) Press Change button.

3) Enter a new password that matches the rule.

4) Press Change button.

5.2 The layout

Once you login in successfully, you can see the screen as below.



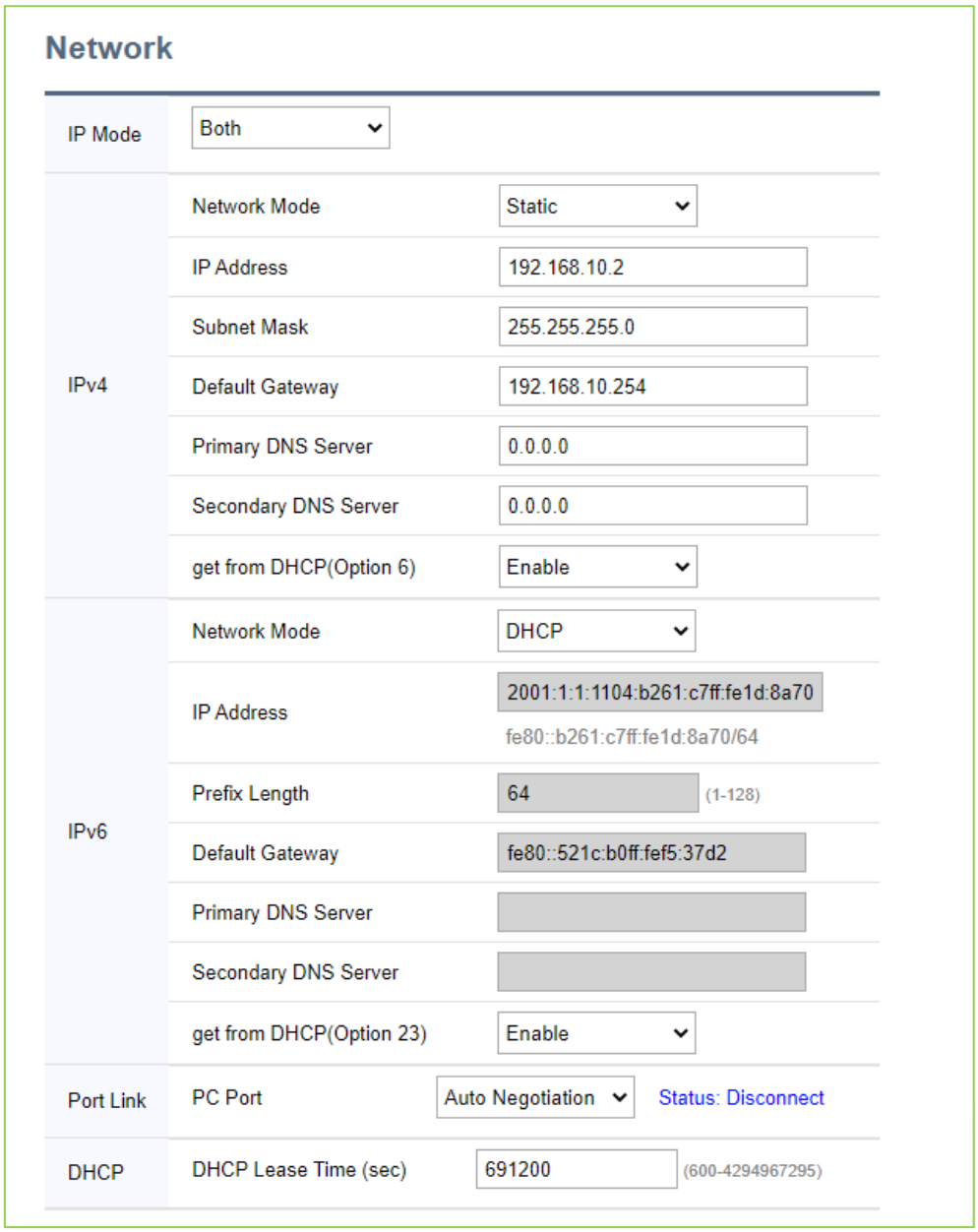
On the left side is the web manager page-link. In the main screen, you can read or set the specific configuration related the page-link you've clicked. The session of the Web Manager is automatically terminated after a certain period of time without any operation. To be away, you must log out by clicking the icon in the upper right corner.

5.3 Configuration management

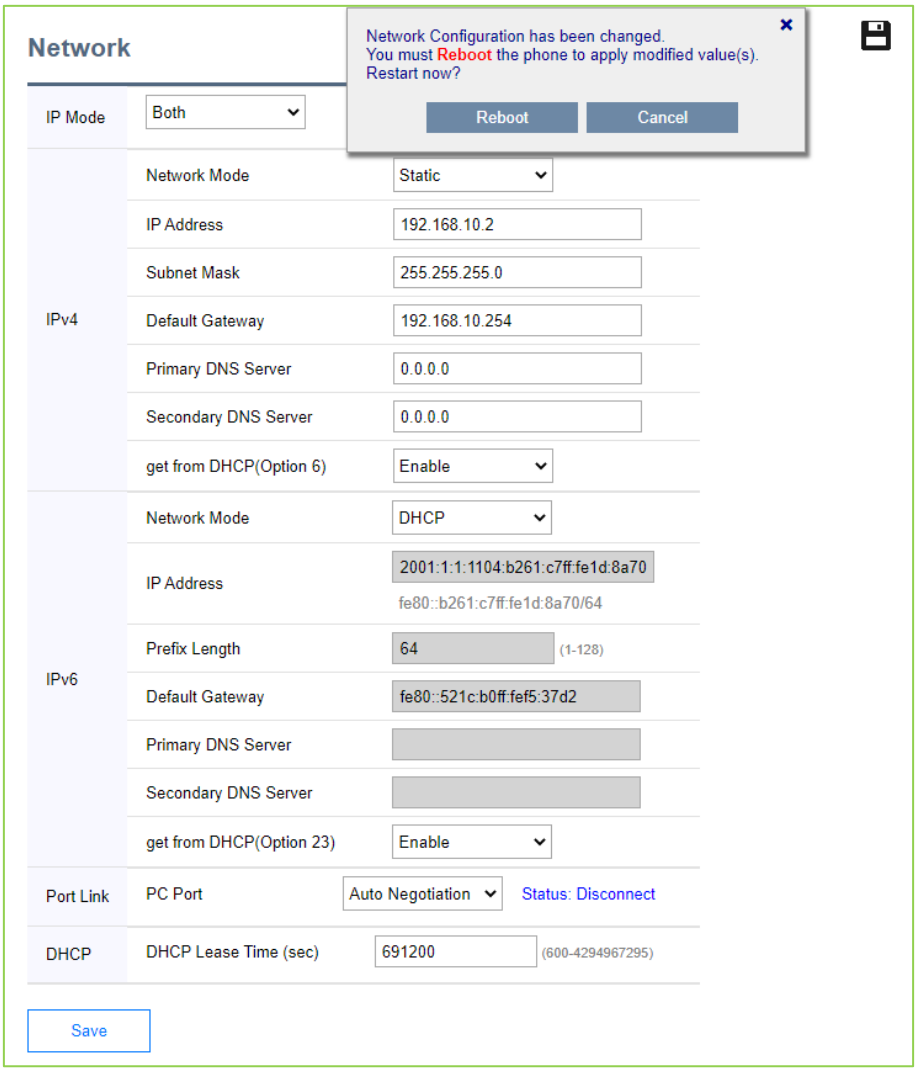
Next figure is about “Network” settings. When you click <Network> link in the left side, all configurations related <Network> are shown in the main screen like a following picture.

Configurations on “Network setting” in the following picture are defined in <Network> node of XML configuration file.

Appendix C. Using the XML configuration file describes all configurations for 1000i phone and their attributes, valid range and a default value.

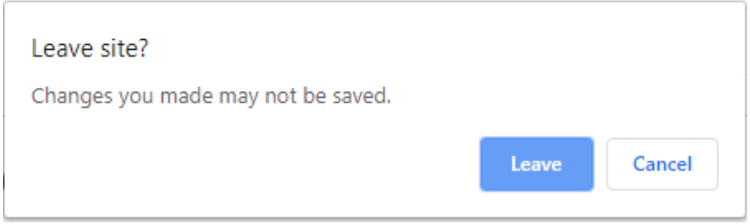


Once you change the value of any field, you must click the save button or the save icon to keep new settings. If it needs to restart for applying new values, a notice appears as below.



You can apply the new settings immediately by pressing the Reboot button on the notice. Alternatively, after pressing the Cancel button and completing the configuration for another node, you can finally reboot to apply the new configuration. Even if you leave the page and configure other nodes, you may not forget to reboot because of the "Reboot required" text at the bottom left of the page.

If you leave the page without saving, a confirmation message like following appears. You can keep the changed in a page and stay there by clicking “Cancel” or leave the page without saving by clicking “Leave”.



The detailed description of each field on the screen is provided in the following sections, and see Appendix C for attributes that are not in this chapter.

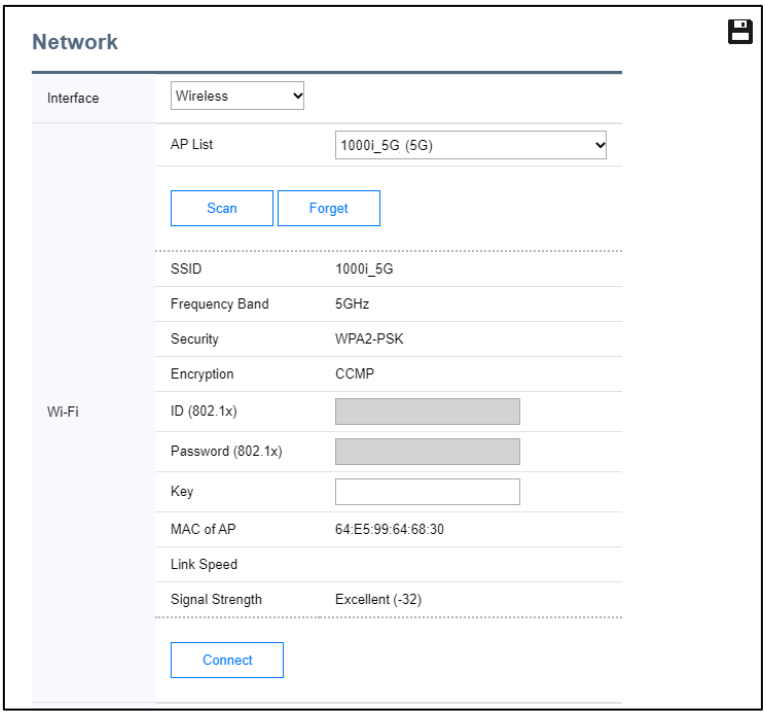
5.3.1 Network

The adjustable parameters in Network menu are:

- IP Mode: IP Phone supports IPv4 and IPv6 both. To use only IPv4, select the “IPv4 ” from this menu. The phone will set only IPv4 to the network interface. To use only IPv6, select the “IPv6”, and to use both IPv4 and IPv6, select the “Both”.
- Interface is the selection of wired network or wireless network when a Wi-Fi dongle is attached. (1040i and 1050i only)
- Network Mode is the option to use DHCP or static IP address.
- IP Address is the IP address of this IP Phone.
- Subnet mask is the subnet mask address of this IP Phone.
- Default Gateway is the default gateway IP address of this IP Phone.
- Primary DNS Server is the DNS server address of this IP Phone.
- Secondary DNS Server is the auxiliary DNS server address.
- Get from DHCP (Option 6) is for getting the DNS server address from a DHCP server by the option #6.
- Prefix Length is the length of the phone's subnet prefix for IPv6.
- Get from DHCP obtains the DNS server address through option number 6 from DHCP server for IPv6.
- PC Port determines the network speed of PC port and whether it is used or not.
- DHCP Lease Time defines the lease time of DHCP request. The possible values are 600 to 4294967295.

To connect using Wi-Fi dongle:

The 1040i and 1050i support the Wi-FI dongle. If Wi-Fi dongle is attached to the phone, you can see the “Interface” menu in the top of the “Network” page.



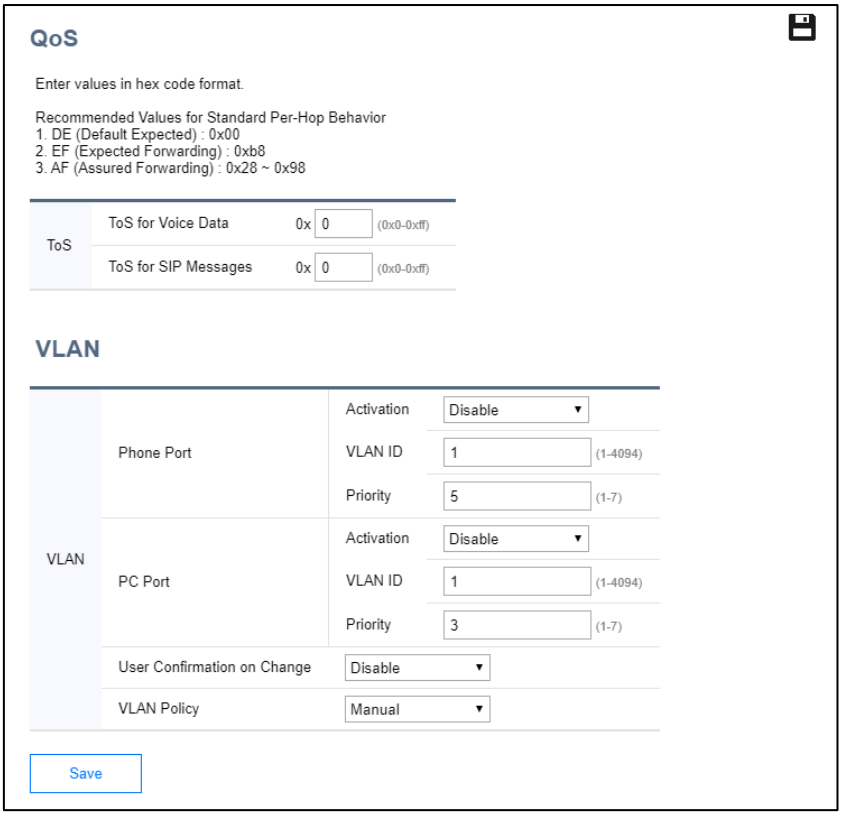
1) Change it to “Wireless”, then the Wi-Fi settings will be shown.
2) Click the “Scan” button, and wait for AP scanning.
3) Select an AP to connect from the “AP List”.

4) Enter “ID” and “Password”, or “Key” according to the connecting AP.
5) Click the “Connect” button, and wait for connecting.
6) Configure the other network settings (Network Mode, IP Address, Subnet Mask, Default Gateway)
7) Click the “Save” button, and reboot the phone.

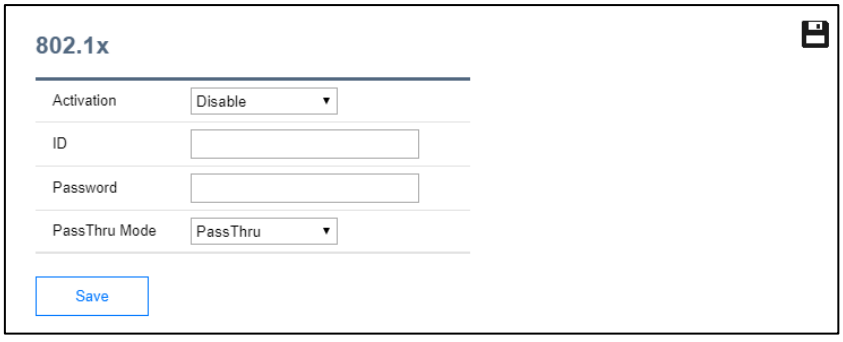
5.3.2 QoS

On the QoS screen, the DiffServ Code Point (DSCP) and VLAN parameters are defined. DSCP or ToS establishes the priority of voice packets. Compatible routers apply the DSCP for packet priority, handling higher priority packets first. However, at congested routers, higher priority packets are discarded first. Thus, a median value for DSCP is recommended.

If LLDP is enabled and VLAN Policy is LLDP, manual VLAN settings are overwritten during a reboot of the IP Phone. To use manually defined VLAN settings, disable LLDP or set the VLAN Policy to 'Manual', see 802.1x and LLDP-MED configuration.



5.3.3 802.1x

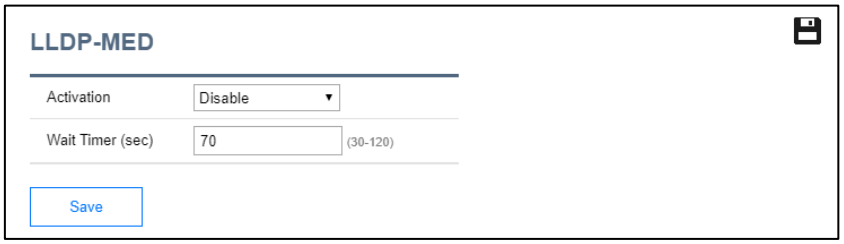


The adjustable parameters in 802.1x menu are:

- Activation enables or disables the response as a supplicant to challenges.
- ID is the username used for 802.1x authentication.
- Password is the password used for 802.1x authentication.
- Passthru Option is the option of passthru for 802.1x authentication.

See section 4.5.9 802.1x for details on 802.1x operation.

5.3.4 LLDP-MED

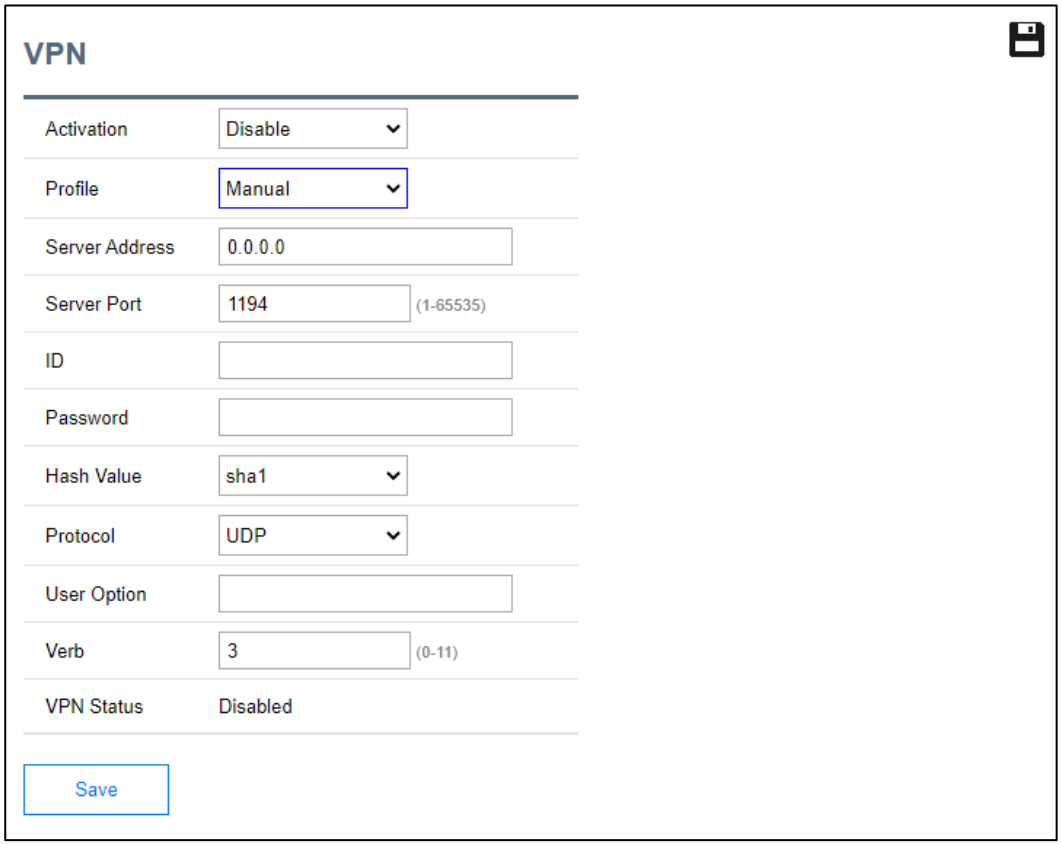


The adjustable parameters in LLDP-MED menu are:

- Activation enables or disables LLDP of the current used profile.
- Wait Timer sets the transmission period of LLDP package.

See section 4.5.10 LLDP-MED for details on LLDP operation.

5.3.5 VPN

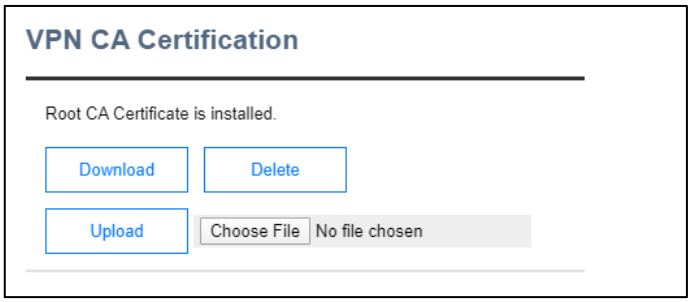


The adjustable parameters in VPN menu are:

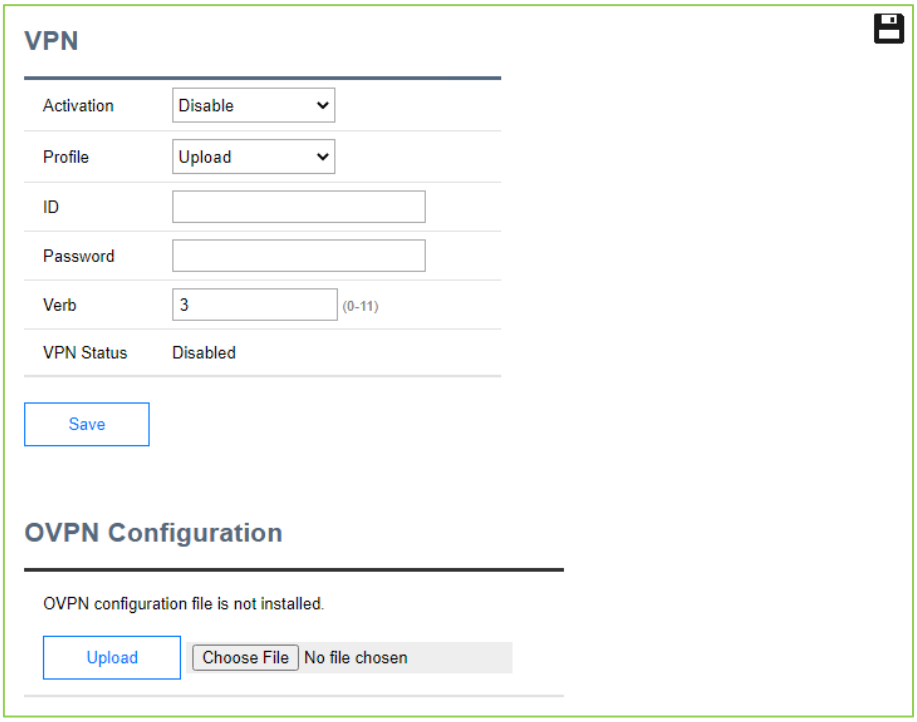
- Activation enables or disables the VPN operation.

- Server Address is the IP address of FQDN of the VPN server.
- Server Port is the port number of the VPN server
- ID is the authentication username for VPN connection.
- Password is the authentication password for VPN connection.
- Hash Value is the hash function algorithm.
- Protocol is the connection protocol with VPN server.
- User Option defines the option values for connecting to the VPN server.

'VPN Status' shows the current status of the VPN operation.



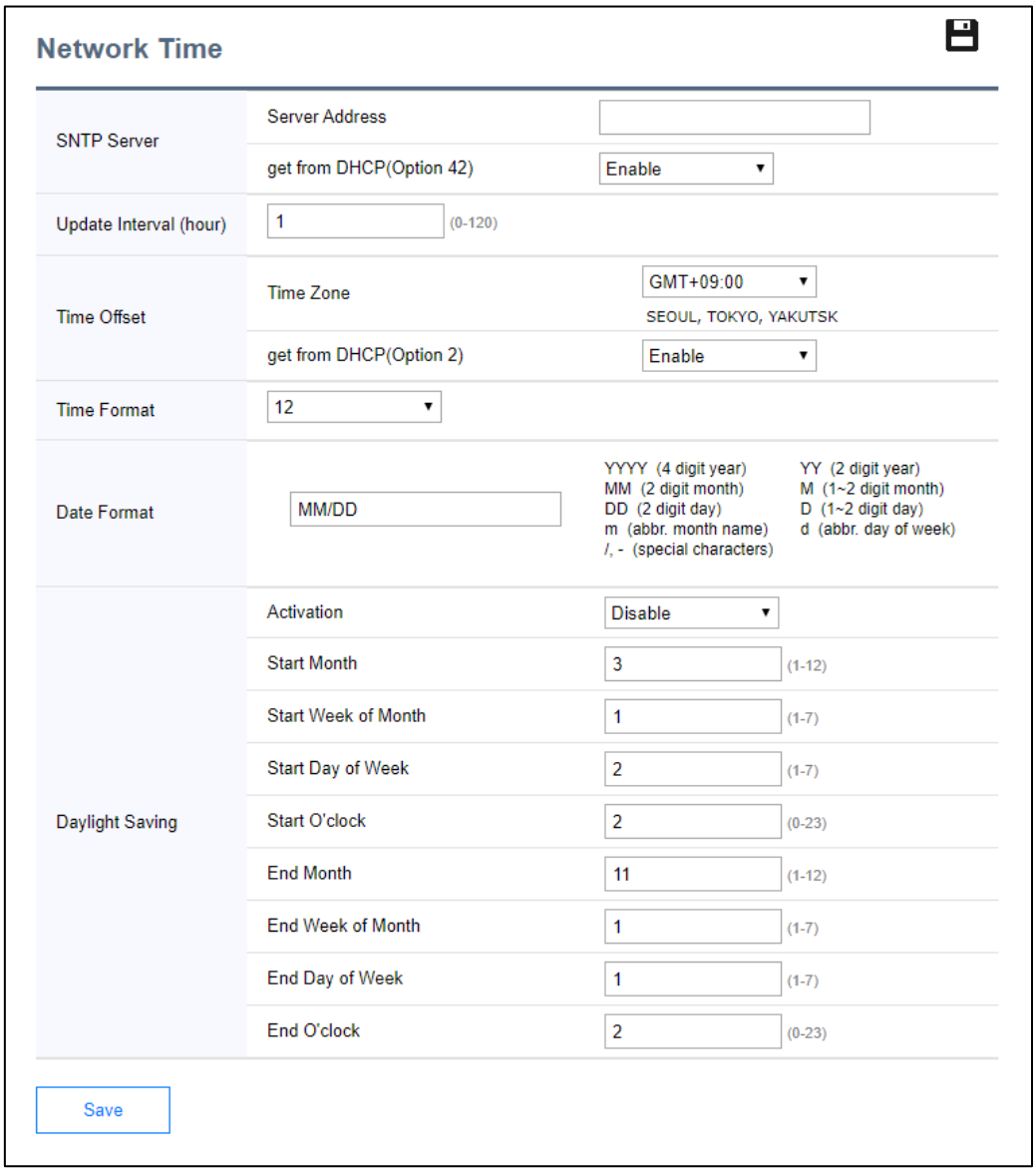
VPN menu has the submenu for managing the certificate file. If there is no Root CA certificate file, you can upload a new certificate file. If there is a certificate file, you can download or delete it.



To use VPN with the OVPN file, change Activation to Enable, set Profile to Upload menu and upload the OVPN file received from the system administrator. If OpenVPN user ID/Password are required, enter the corresponding value. After uploading all necessary files and completing all settings, press the Reboot menu on the left to restart the phone and apply the new settings.

5.3.6 Network Time

Network Time screen allows you to set the configuration for time and daylight savings time.



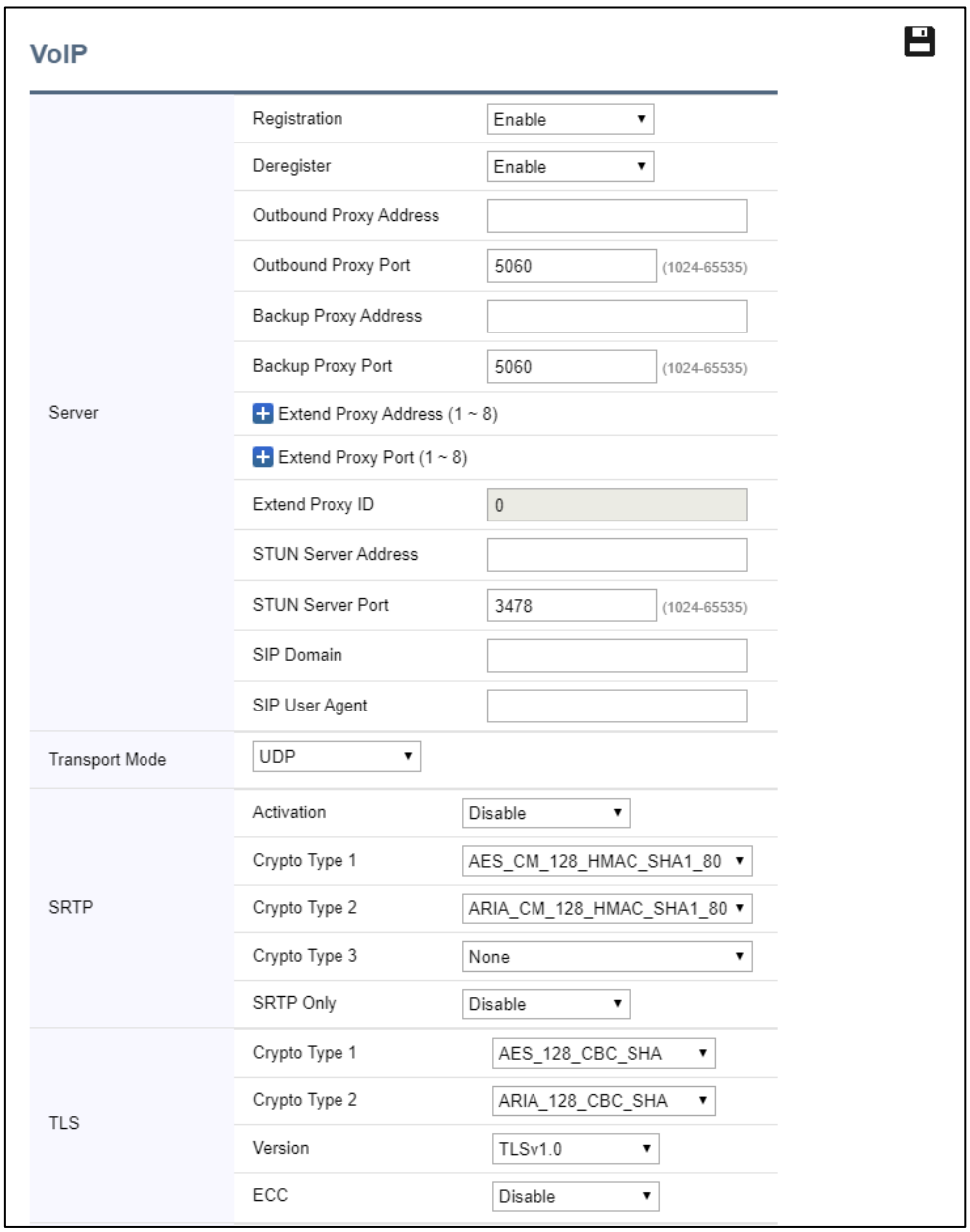
The following parameters are set in Network Time configuration:

- SNTP Server Address is the IP address or FQDN of the SNTP(Simple Network Time Protocol) server.
- Update Interval
- Time Offset defines the local time zone of the IP Phone.
- Time Format sets the format of the LCD time display; OFF, 12-hour or 24-hour format.
- Date Format sets the format of the LCD date display
- Daylight Saving section defines the settings of DST in detail, when DST is enabled.

Well-known SNTP servers are:

- ntp.nasa.gov
- clock.via.net
- tick.ucla.edu

5.3.7 VoIP

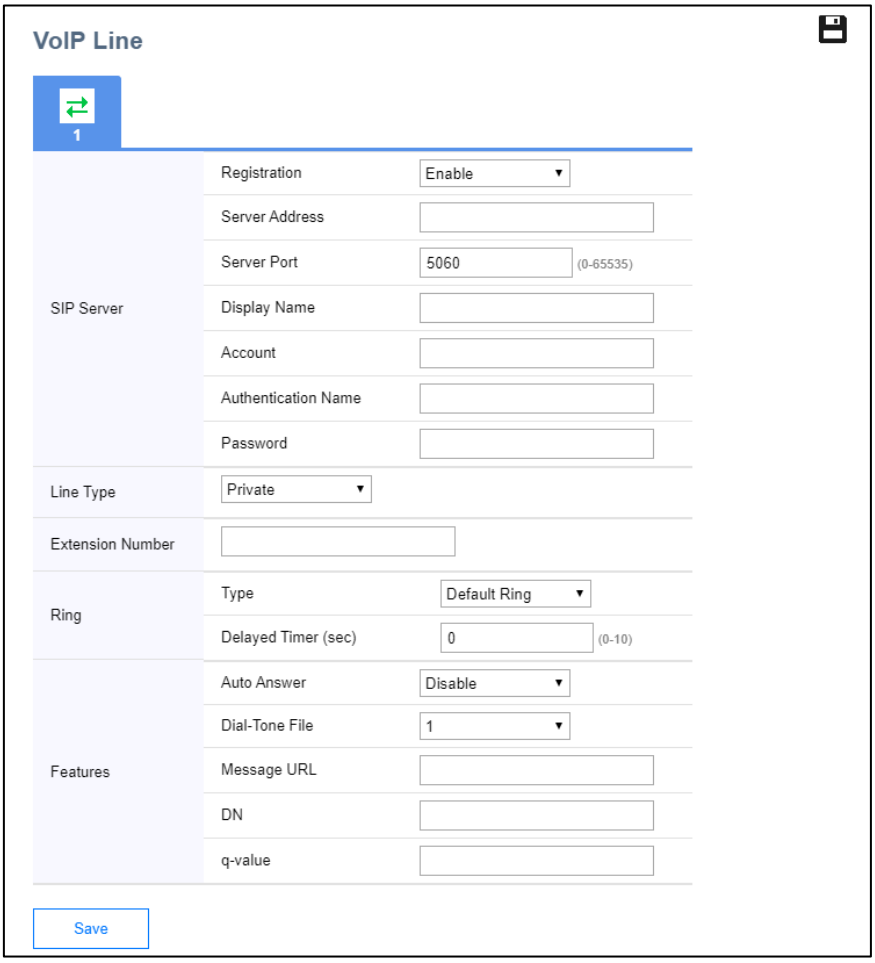


The following parameters are set in VoIP menu :

- Outbound Proxy Address is the IP address or FQDN of the outbound proxy server to receive SIP requests in place of the line-based Proxy Server.
- Outbound Proxy Port is the port of the outbound proxy server.
- Backup Proxy Address is the IP address or FQDN of the backup proxy server, used when the connection with the primary proxy times-out.
- Backup Proxy Port is the port of the backup proxy server.
- STUN Server Address is the IP address or FQDN of STUN server.
- STUN Server Port is the port of STUN server.
- Transport Mode is the transport type (UDP, TCP) of sending and Receiving SIP packets.
- RTP Start Port is the port for Real-time Transport Protocol (voice) packets.
- Codec selects the priority for audio codecs.

See Appendix C for other parameters.

5.3.8 VoIP Line



There are as many tabs as the number of configured voip lines in the table. Selecting a tab shows ameters to set for that VoIP Line.

The following parameters are set in VoIP Line configuration:

- Server Address is the IP address or FQDN of the SIP call server or SIP proxy.
- Server Port is the server port for incoming requests and outgoing responses.
- Display Name is the name used as part of the IP Phone caller ID. The display name assigned for Line1 displays on the LCD display.
- Account is the name used during registration. The name is generally the telephone number or the SIP user Id.
- Authentication Name is the user id for authentication of the line appearance with the call server.
- Password is the password for authentication of the line appearance with the call server.
- Line Type is the type of line (private, shared (BLA), or DSS line).
- Extension Number for DSS is the DSS number when the type of line is the DSS.
- Ring Type sets the ring tone for incoming calls on the line. The Ring Type can be "Default Ring", "Ringer Off" or Ring type 1 to 4, Wave Ring 1-8 or Wave Ring 1-4 depending on a model. The default ring is configured under the Sound Preference.
- Delayed Timer delays audible notification (ring) for a line. The delay is set in 1-second increments from 0 (no delay) to 10 seconds.
- Auto Answer is the flag of the auto answering when a line has an incoming call.

See Appendix C for other parameters.

5.3.9 Dial Plan

The IP Phone incorporates a dial plan to monitor if the number the user dialed matches the dial plan number. If the call matches, it modifies the number and handles the call using the instructions in the dialing plan.

The Digit Map Table allows you to define a dial plan by entering a string of up to 512 characters. You can enter individual numbers within a string, separated by '|'.

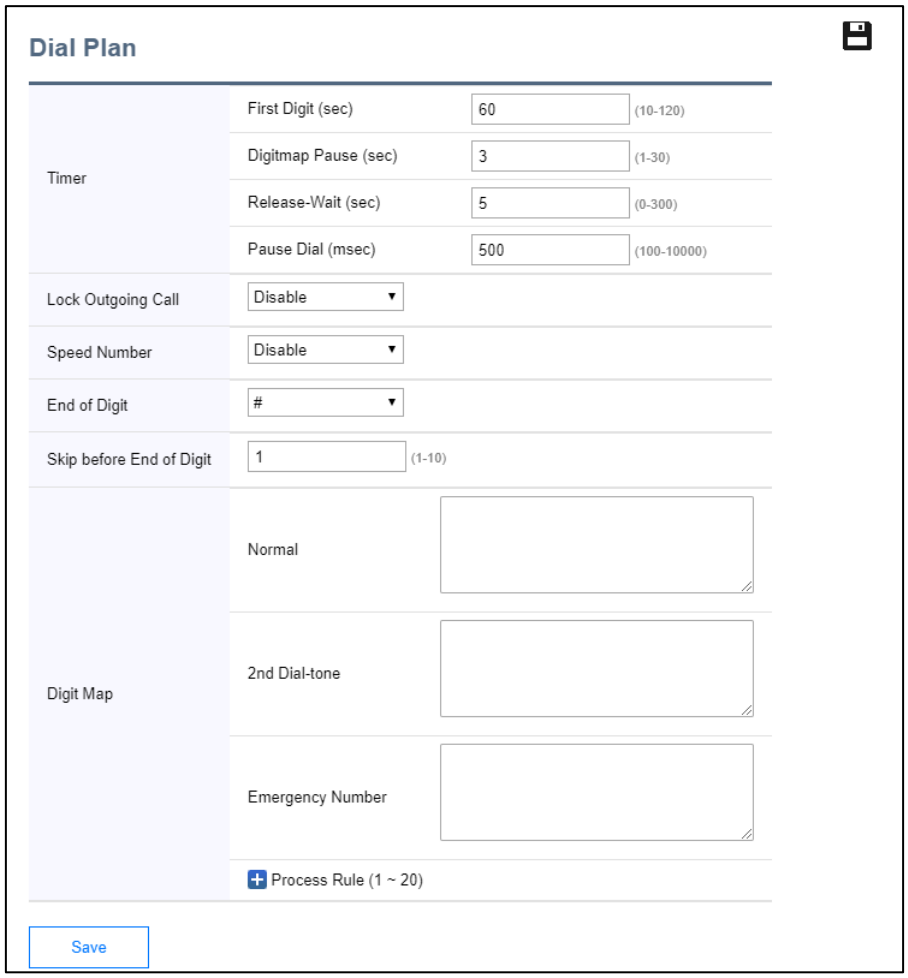
In addition to dial pad seats, special characters can be entered as follows:

- An ‘x’ matches any single digit, (0 to 9).
- A ‘T’ inserts a timed pause for additional digits.
- A ‘.’ matches any number of occurrences of a digit, including zero occurrences and includes special character matches.
- A ‘+’ matches any number of occurrences of the previous match, including special characters.
- The ‘[ ]’ matches a digit from a group of digits. This character combination is used with “.” to define a range, [2-6] matches any digit from 2 to 6 inclusive.

The dial plan can include modification of a matching dialed number before call set-up. The Digit Map Process defines the modification procedures for a number.

The process may include:

- TRNC defines the number of digits to delete from the dialed number.
- Pre defines digits to insert in the front of the truncated number.
- Post defines the digits to insert at the end of the truncated number.



The following are examples of Dial Plans using the digit map tables.

Simple Dial Plan

- Customer requirement: Process the call after any 7 digit number is dialed or if the Operator is dialed digit ‘0’.
- Digit map routing entry: The digit map of this Dial Plan is (0T|xxxxxxx).

From this map, if the user dials '0' as the first digit, the IP Phone will wait for the pause time for additional digits. After the pause, if no additional digits are dialed, the phone places a call to the Operator. In addition, after the user dials any seven digits, the call is processed.

Complex Dial Plan

- Customer requirement: Process the call after the local operator at 0, long distance operator at 00, a four-digit local extension number starting with 3,4 or 5, a seven-digit local number prefixed by an 8, two-digit number preceded by star (‘*’), a ten digit long distance number prefixed by 91, or an international number starting with 9011 is dialed.
- Digit map routing entry: The dial plan for this is (0T|00T|[3-5]xxx|8xxxxxxx|*xx|91xxxxxxxxxx|9011x.T).

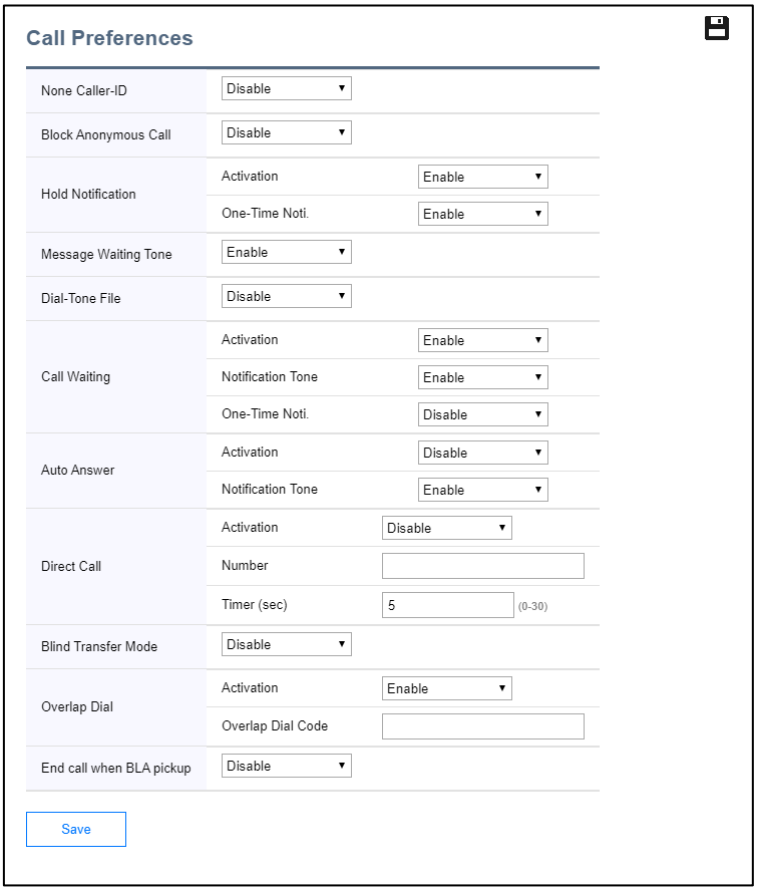
2nd Dial Tone

A second 512-character string map is employed to provide 2nd Dial Tone.

- Customer requirements: To provide second dial tone after a ‘9’ is dialed, establish a Dial Plan as below.

In the Digit Map (2nd Dial Tone) text box, the dial plan for this is (9|).

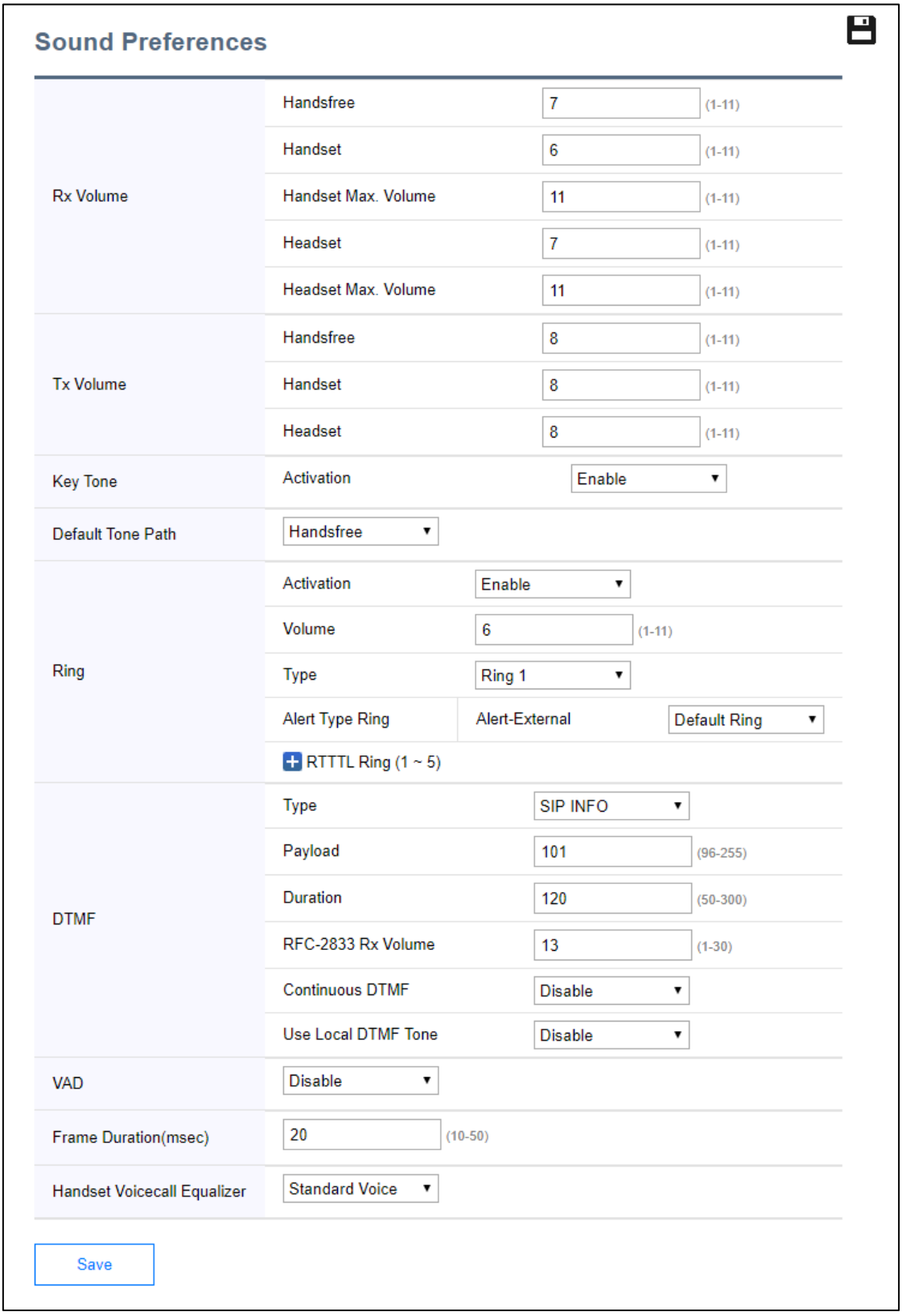
5.3.10 Call Preferences



- Max. Number of Call defines the maximum number of active calls on one line and defines the total number of calls in a phone.
- None Caller-ID Blocking enables and disables blocking the outgoing caller id. When enabled, the IP Phone sends the anonymous id.
- Block Anonymous Call, when enabled, blocks incoming calls with an anonymous caller id.
- Hold Notification, when enabled, rings the phone if the user returns to idle with a call on hold.
- Message Waiting Tone, when enabled, stutter dial tone notifies the user a new message is waiting. Stutter dial tone notification is in addition to the visual flashing Message LED notification.
- Call Waiting Function, when enabled, visually notifies the user of an incoming call while in a conversation. If disable, when in a conversation, a new call is refused, and the call server routes the call.
- Call Waiting Tone, when the IP Phone is in use with call waiting enabled, a beep tone delivered over the existing call indicates an incoming call.
- Auto Answer, when enabled, will activate the speakerphone or headset automatically if an incoming call is received.
- Direct Call, when enabled, places a call to a predefined destination automatically when the phone goes off hook.
- Direct Call Number defines the destination for a direct call.
- Direct Call Timer sets a delay timer for Warm-Line operation. The timer specifies the period before placing a direct call. The timer is set in 1 second increments from zero (0) to ten (10) seconds. For Hotline operation, set the timer to zero (0) seconds.

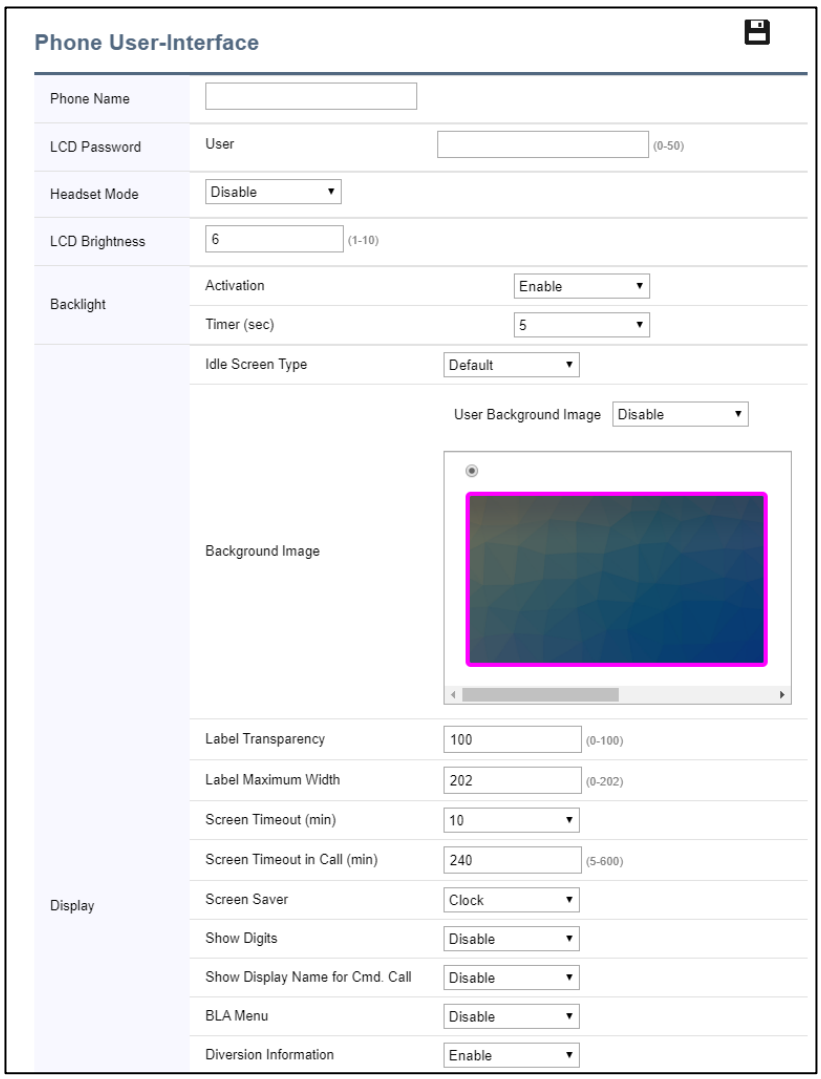
See Appendix C for other parameters.

5.3.11 Sound Preferences



See Appendix C for other parameters.

5.3.12 Phone User-Interface

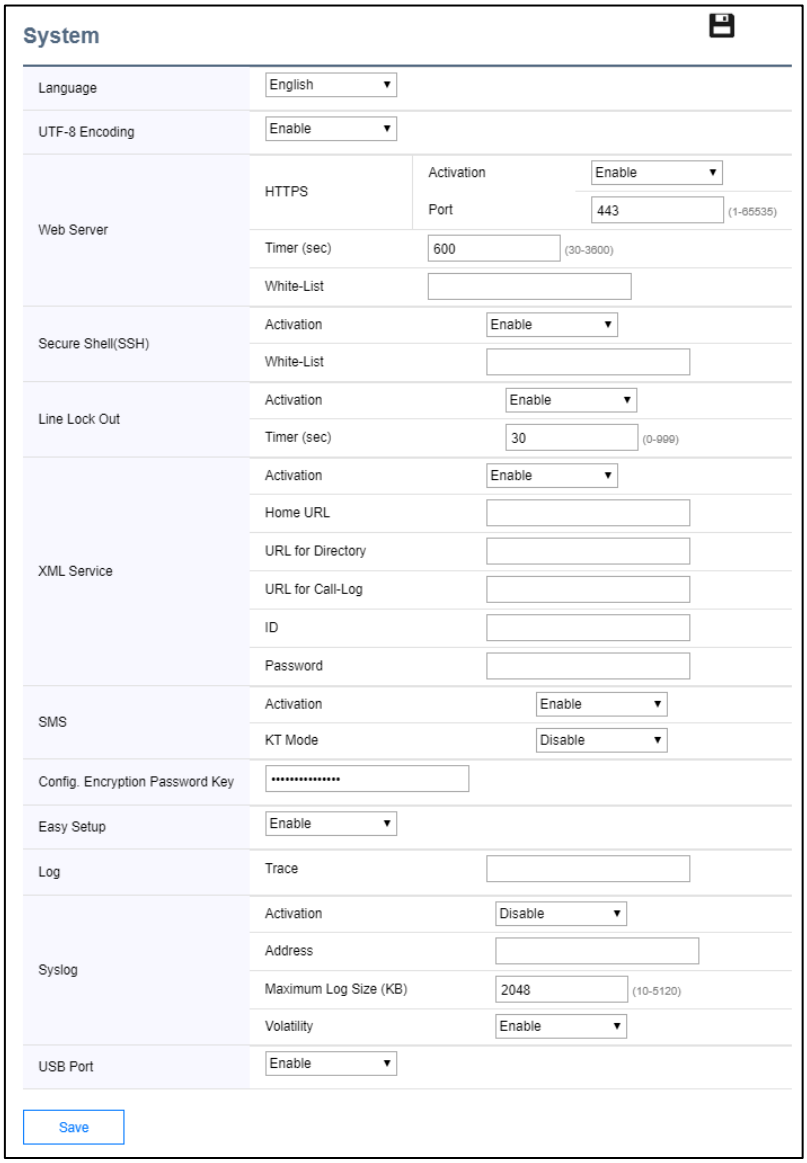


The adjustable parameters in Phone User-Interface menu are:

- Phone Name is shown on the top of the phone LCD.
- LCD password can be differently set for Admin/ User.
- Headset Mode enables headset operation.
- LCD Contrast set the contrast level for the phone's LCD. (for 1010i, 1020i)
- LCD brightness set the level of the contrast of LCD. (for 1030i, 1040i, 1050i)
- LCD dimming, when enabled at 1020i, the LCD will turn on and stay slightly dimmed.
- Backlight is a setting about turning on the backlight when the terminal is idle.
- 1030i, 1040i, and 1050i allow you to customize the background image of the IDLE screen using Background Image menu. Note that the appropriate image size varies by the device model. As image type, BMP or PNG file can be applied.
- Label Transparency allows you to set the opacity level of the IDLE screen.
- Return to Active Call, sets the timer to return to the active call from a waiting call, a held call, and etc on the LCD display. If you receves an incoming call during conversation, the phone will display the incoming call on the LCD, and it will display the active call again on the LCD after this timer. If you set it to 0, the phone will not return to the active call.

See Appendix C for other parameters.

5.3.13 System



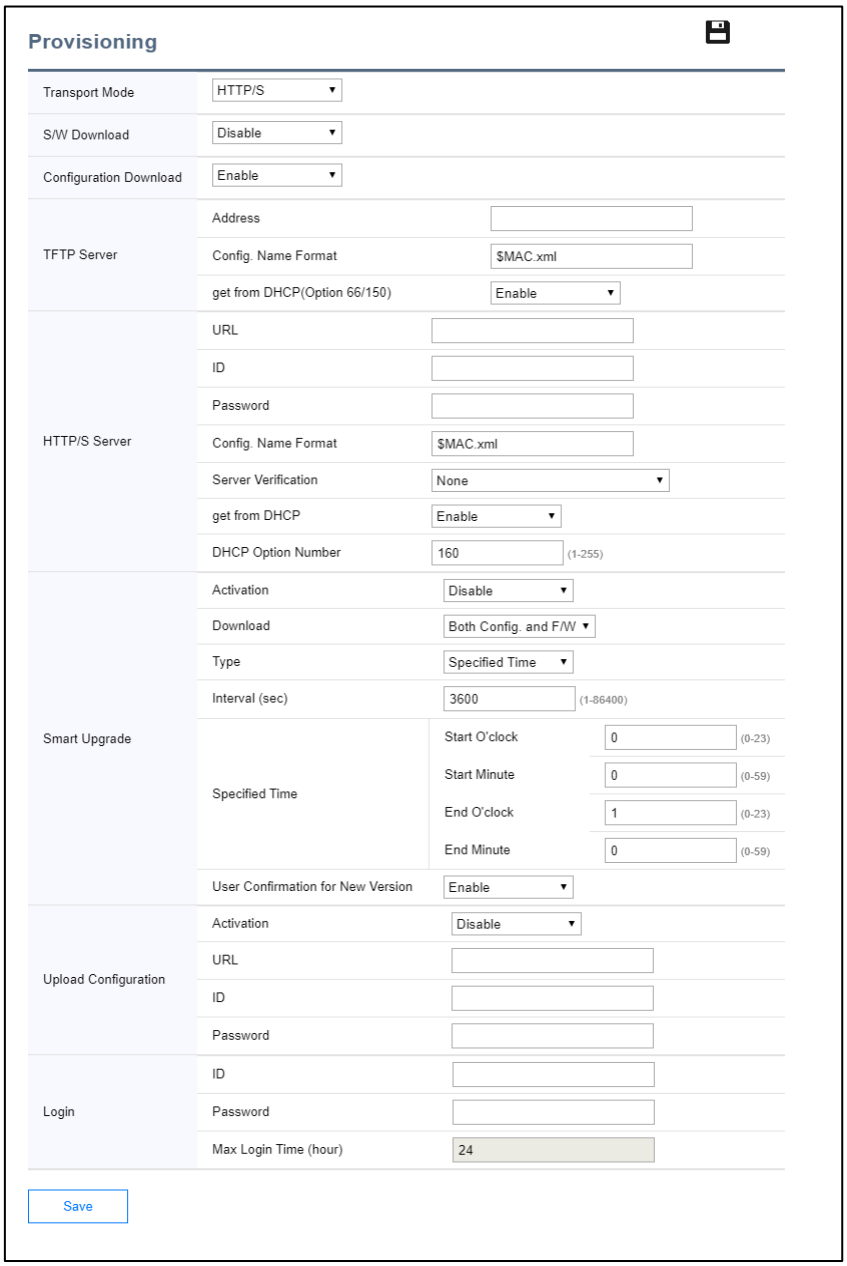
The adjustable parameters in System menu are:

- Web Server, when enabled, the web manager is available, and if there is no action, the session will expire after the time defined in the Timer.
- Web server's Whitelist has the address list that can be accessed by the web manager.
- Line Lock Out, enables LLO feature, and starts LLO after the time defined in Timer.
- Log switches the trace flag to gather log from the IP Phone.
- Syslog saves the log to the IP Phone internally, or to an external syslog server.
  - Activation: enable/disable syslog feature
  - Address: IP address of an external syslog server. When empty, it saves internally.
  - Maximum Log Size: limitation of the log
  - Volatility: Whether the log is saved to the flash memory or the RAM. If it is set to "Enable", the log will be lost after rebooting the IP Phone.
- Enable/Disable USB port (for 1040i, 1050i)

See Appendix C for other parameters.

5.3.14 Provisioning

The adjustable parameters in Provisioning menu are:



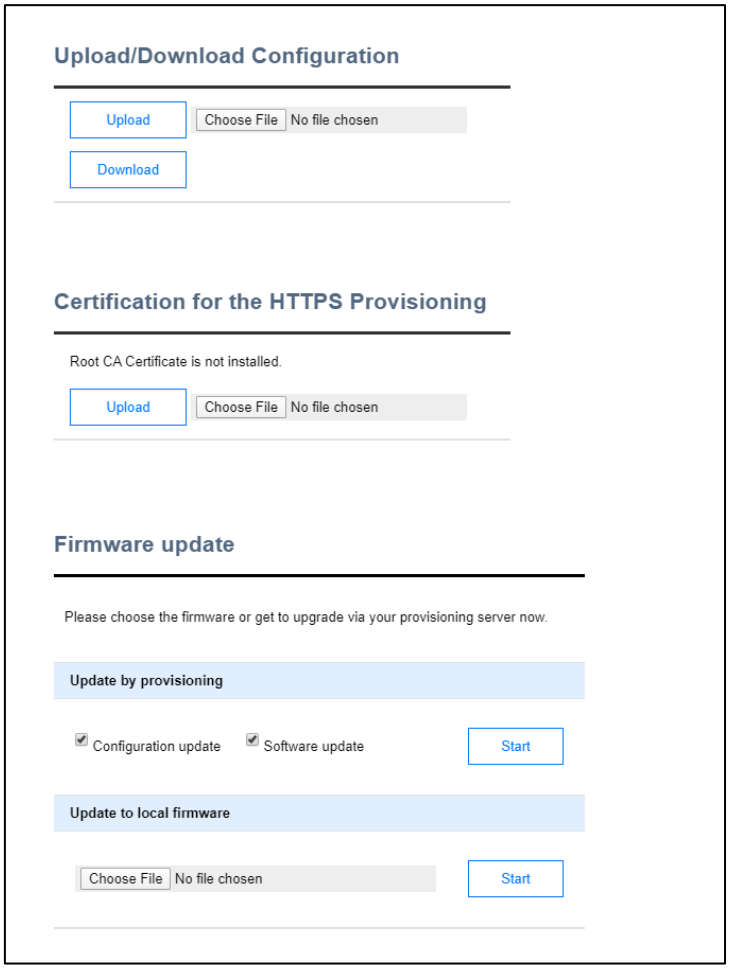
- Transport mode sets the protocol of the server where configuration and software files are stored.
- SW Download enables and disables updating firmware from the provisioning server.
- Configuration Download enables and disables applying new configuration from the provisioning server.
- TFTP Server
- Address assigns the IP address or FQDN of the TFTP server where configuration and software files are located.
- Config. Name Format defines the format of the MAC (Phone specific) configuration filename; “$MAC” is replaced with the MAC address of the IP Phone and must be included as part of the file name or path to the configuration file on the TFTP server
  - HTTP/S Server

- URL is the address of the provisioning server. When HTTP/S is selected for Transport Mode, SW or configuration is downloaded from the serer at this address.
- ID assigns the username for HTTPS authentication.
- Password sets the password for HTTPS authentication.
- Config. Name Format defines the format of the MAC (Phone specific) configuration filename; “$MAC” is replaced with the MAC address of the IP Phone and must be included as part of the file name or path to the configuration file on the HTTP/HTTPS server
- Get from DHCP enables the IP Phone uses the DHCP Option defined to retrieve the URL. If the option is present, the provisioning type is changed automatically to HTTP/HTTPS using the URL from the DHCP option, default 160.
- Smart Upgrade enables periodic checks for new software on the provisioning server. Two methods of smart upgrade are available; one checks the server at the interval in hours, the second checks daily at the time specified in a 24-hour clock format.
- If the IP Phone finds a new version of software during the smart upgrade, the IP Phone either downloads the file or displays the availability of the new version and awaits user confirmation to download.

See Appendix C for other parameters.

The provisioning menu has a submenu as shown below.

If a provisioning server and protocol are specified, you can upload or download the configuration, or run the provisioning process immediately. In addition, if there is a firmware file of the IP Phone, the firmware can be applied directly to the phone without using a server by menu - Firmware update.



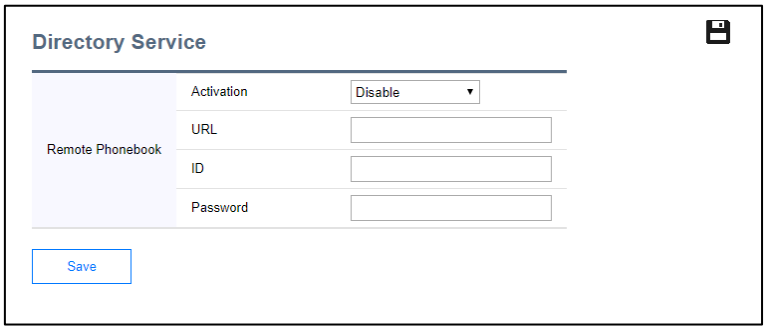
5.3.15 Programmable keys

1000i has several flexible buttons, and you can assign buttons that are not used on a line to various functions. In addition, expansion modules can be connected to the 1000i phones to extend the number of flexible buttons.

See Appendix A, for the number of buttons available for each model.

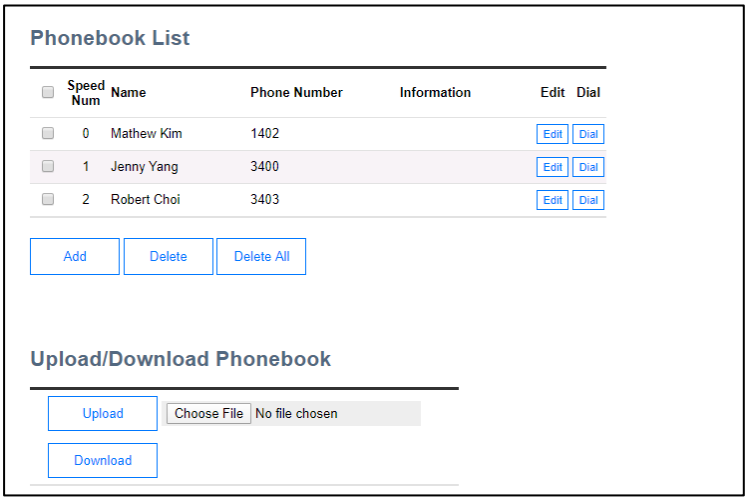
The buttons on the screen are arranged in the same way as the physical location of the actual terminals. When you set a button's properties, the fields you can input differ depending on the value you select from the drop-down menu, and the fields that require input are distinguished by the background color.

5.3.16 Directory Service



See Appendix C for other parameters.

5.3.17 Phonebook



IP Phones have an internal phonebook with 1000 contacts. Calling a contact may use the Speed number (0-999), depending on your options. The contact number or SIP user id is maintained in the Phone Book.

You can add, edit, and delete records in the Phone Book menu. 100 records are displayed per screen page. The phone book can be downloaded or uploaded as a * .csv format file.

To Upload

1) Click [Choose File] and locate the desired csv file in you PC.
2) Click [Upload] button.

To Download

✓ Click [Download] button, you can get phonebook.csv file.

To Add a new contact

1) Click [Add] and fill in the required fields.
2) Click [Save] button.

To Delete contacts

1) Select the checkbox at the beginning of each line in the table, then press the delete button to delete the selected entries.
2) Selecting the checkbox in front of 'speed num' in the table header selects all the data in the table.

5.3.18 Call Lo

Call Log is a list of all calls (incoming, outgoing or missed) at the extension. The log shows the call type (IC, OG and M), the contact name or number, contact information, time and duration of the call, and the call status.

You can call a contact from the call log or save the contact record directly in the phonebook.

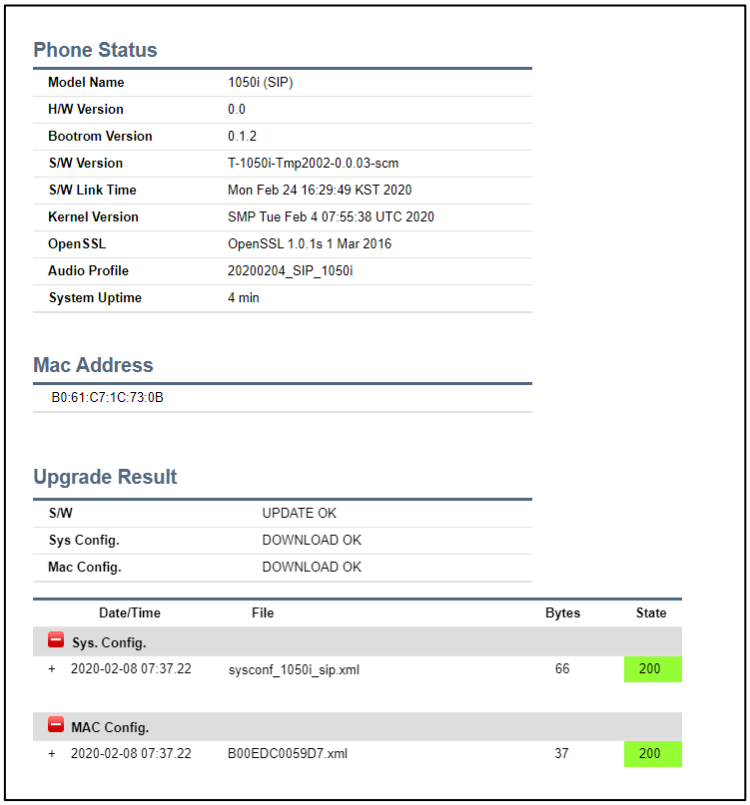
To Add a new contact from the Call Log

1) Click [+] button.
2) Fill in the required fields.
3) Click [Save] button.

To call a party from the Call log:

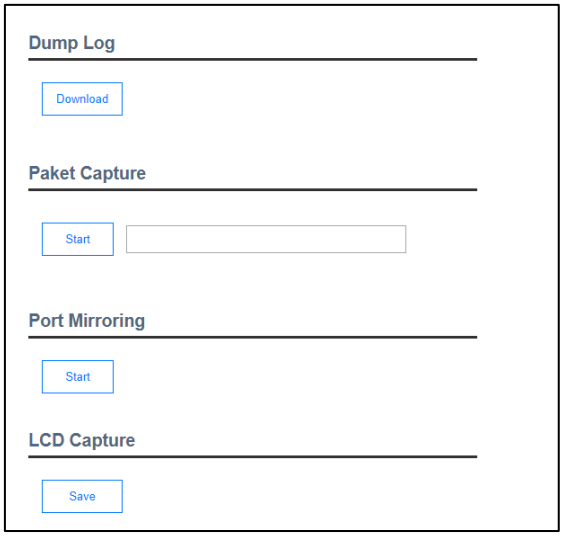
✓ Click the [Dial] button associated with the desired Call log record.

5.3.19 Phone Status



The Phone Status screen shows detailed version information for the 1000i phones, the MAC address of the phone, and the results of the upgrade process.

5.3.20 Diagnosis



To check the phone, some diagnosis tools are provieded in the “Status” > “Diagnosis” menu.

To dump a log:

1) Press the “Download” button in the “Dump Log” section.
2) Wait for a while.
3) Log file is saved to the PC.

To start capture the packets from/to the phone:

1) If you want to capture some specified packets, enter a filter string in the " Packet Capture" section.
2) Press the "Start" button.
3) The phone will start the capturing of packets.

To stop capture the packets and download:

1) Press the "Stop" button in the "Packet Capture" section.
2) Press the "Download" button.
3) Packet captured file is saved to the PC.
4) Press the "Delete" button to remove the temporary log file saved in the phone.

To start port mirroring from the LAN port to the PC port:

1) Press the "Start" button in the "Port Mirroring" section.
2) The PC connected to the PC port of the phone can capture the packets from/to the phone.

To stop the port mirroing:

✓ Press the "Stop" button in the "Port Mirroring" section.

To capture a LCD screen:

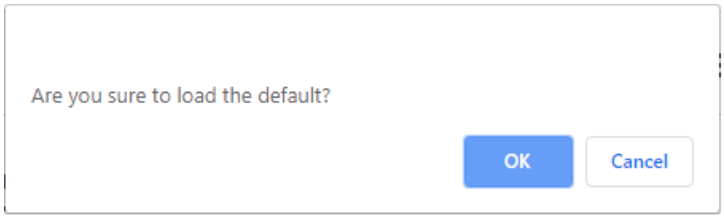
1) Press the "Save" button in the "LCD Capture" section.
2) Captured image file is saved to the PC.

5.3.21 Set to default

Set to Default menu allows IP Phone to return to factory defaults. See Appendix C for more information.

To set the IP Phone memory to the default configurations:

1) Select [Set to Default] menu in the left.
2) Confirm to load the default, then the IP Phone loads the default configuration and reboots.



6 Troubleshooting

6.1 Information

The 1000i phones provide the means to quickly view the Network parameters (IP address, subnet-mask, and default gateway address), server IP addresses (call server, TFTP server, and SNTP server), and results of the last file download attempt.

1) Press the Menu button.
2) Select “Status” menu.
3) Select “Information” menu.

The download information displays results of the most recent download attempt, see Appendix B. Download LCD Messages for a list of the messages.

6.2 Trouble Shooting Guide

Table 6.2-1: Troubleshooting

Symptom	Corrective Action
LCD Off	• Switch power source ON • Check/replace AC/DC adapter • Check/correct input power from the LAN
Handset off-hook, no dial tone	• Assure the RJ-45 LAN cable is connected correctly. • Make sure that the call server IP Address is correct.
Button LEDs flash red then orange	Line not registered with the call server, • Check phone configuration • Check network connections • Check call server.
Phone displays “DHCP Searching” constantly, more than 2 minutes.	DHCP server is not responding. • Check DHCP server. • Check VLAN settings. To exit the DHCP Searching mode, press Menu button 3 times, the phone will complete the boot and return to idle.
Phone displays “VLAN Changed. Reboot?”	The downloaded configuration file has changed the VLAN settings. After the next reboot, the phone will use the new settings. Select Yes to reboot now using the new settings. Select No, the Phone will not use the changed VLAN settings until the next reboot.
Network setting error	**Network Mode: DHCP** • Make sure Network Mode is DHCP. • Make sure the RJ-45 LAN cable is connected securely to your IP Phone and LAN, switch, or router. • Make sure that all network ports, routers, or switches are working. • Make sure your network supports DHCP.

Symptom	Corrective Action
	Network Mode: Static ・ Make sure Network Mode is static. If you entered the IP Address through a phone, make sure you reset your IP Phone.
When I attempt to make a call, I can't hear	・ Check all cables and connections. ・ Make sure the call server IP Address is correct.

Appendix A: 1000i Series Model Specification

Enjoy a wide selection of easy-to-use system IP Phones to enrich your communications experience with a powerful speaker phone and wideband voice support.

1000i Series phones from Entry level to advanced terminals, designed to best fit the users' needs.

Table A: 1000i Series Models Specification

Specification		1010i	1020i	1030i	1040i	1050i
Display	LCD Size	2.4"	2.8"	2.8"	3.5"	4.3"
	LCD Resolution (Pixels)	132 x 64	132 x 64	320 x 240	480 x 320	480 x 272
	UI Line	4 Line	4 Line	6 Line(Max)	6 Line	8 Line
	Backlit	N/A	Yes			
	LCD Type	Black and White	Gray scale graphic	Color graphic		
Keypad	Programmable button with BLF	YES (4)	YES (8)	YES (6)	YES (8)	YES (12)
	Softkey	YES(3)				
	Navigation key	Yes				
	Fixed button (dedicated hard keys)	YES (11 buttons)	YES (11 buttons)	YES (11 buttons)	YES (11 buttons)	YES (11 buttons)
	LED Buttons	YES (1 Color LED)	YES (3 Color LED)			
Self labeling		4 buttons	4 buttons x 3 pages	6 buttons x 3 pages	8 buttons x 3 pages	12 buttons x 3 pages
Paper labeling		-	4 buttons	-	-	-
Total program-able keys		4	16	18	24	36
Network interface	Ethernet (IEEE 802.3/1 Gigabit)	YES (2*10/100 Base-T)	YES (2*10/100/1000 Base-T)			
	EHSA Support	N/A	YES			
Wireless interface	USB dongle Support	NO			YES	
Speakerphone		YES (Full duplex)				

Specification		1010i	1020i	1030i	1040i	1050i
Audio interfaces	Handset (RJ-9)	YES				
	Headset(RJ-9)	YES				
	Headset(USB)	N/A			YES	
IM Presence		N/A		YES		
BLF		YES				
Power	Local(Optional - 48V,0.3A)	YES				
	PoE (802.3af)	Class 2	Class 2	Class 2	Class 3	Class 3
Audio	Codec	G.711, G.729AB, G.722, Opus				
	Call Signaling Protocol	SIP protocol				
	Features	Echo cancellation, VAD, CNG, Hearing Aid Compatible				
Network	IP Address	Static / DHCP				
	LLDP-MED, VLAN	YES				
	QoS	Dynamic jitter buffering, 802.1p/Q, Layer 3 TOS, DiffServ				
Security	Standard	SRTP w/AES-128, 802.1x/EAP-MD5(EAPOL), Password login				
	VPN (Open VPN)	YES				
Management	Auto/Remote Upgrade	YES				
	Web Management	YES				
Others	Phone tilting	YES (2 Level Bracket)				
	DSS Support	N/A	YES(1024idss,1048ilss,1048idss)			
	Wall mount	YES				
	Ring type	Wave file/FM method				

Appendix B. Download LCD Messages

Table B-1 provides the messages displayed for the most recent software download attempt.

Table B-1: Software Download Message

Message	Description
LOAD FLAG OFF	S/W load is disabled
NO SERVER ADDR	TFTP / HTTP/S server address is 0.0.0.0
VER FILE XFER ERR	Downloading a new version of S/W has failed Check the TFTP / HTTP/S server
IMAGE XFER ERR	
VER FILE NOT FOUND	The version file doesn't exist in TFTP / HTTP/S server
VER STRING ERR	The version file contains an erroneous format
IMAGE NOT FOUND	The S/W image doesn't exist in TFTP / HTTP/S server
IMAGE SIZE ERR	The S/W image size is 0 or larger than maximum size
ID/PASSWORD MISMATCH	The current Id or password for authentication is invalid
NEW VERSION LOADED	S/W was successfully loaded
UPDATE NOT NEEDED	The current version is the latest

Table B-2 provides messages displayed for the most recent configuration file download attempt.

Table B-2: Configuration Download Message

Message	Description
LOAD FLAG OFF	Config file load is disabled
NO SERVER ADDR	TFTP / HTTP/S server address is 0.0.0.0
CFG XFER ERR	Downloading the configuration file has failed; check the TFTP / HTTP/S server status
FILE SIZE ERR	The configuration file size is 0 or there is an error during transfer from the TFTP server
FILE NOT FOUND	The configuration file doesn't exist in TFTP / HTTP/S server
PARAMETER ERR	The configuration file contains an erroneous value; the value has been automatically converted to previous value
ID/PASSWORD MISMATCH	The current Id or password for authentication is invalid
DOWNLOAD OK	Configuration was successfully downloaded

Appendix C. Using the XML configuration file

During automated provisioning with the Config. Download flag, IP Phones attempt to download a global and a phone specific XML configuration file from a TFTP or Web server supporting HTTP/HTTPS. The global files are typically for parameters with the same value for all IP Phones connecting to the server. Parameters such as Network mode, Provisioning, Daylight savings time mode are typical globally settings. Table C-1 below shows the Global XML configuration file name.

Table C-1: Global XML configuration file name

Model	Global XML configuration file name
1010i	sysconf_1010i_sip.xml
1020i	sysconf_1020i_sip.xml
1030i	sysconf_1030i_sip.xml
1040i	sysconf_1040i_sip.xml
1050i	sysconf_1050i_sip.xml

The phone-specific XML configuration file as a default uses the MAC address of the phone for the file name, for example 00405a123477.xml. When using HTTP/S, the IP Phone can be assigned with a name for the phone specific file in place of the MAC address but the MAC address must be included in the file name or path. Parameters specified in the phone specific XML configuration file override those defined in the global XML configuration file.

Both the global and MAC XML configuration files can configure any of the parameters of the IP Phone; the only difference is the MAC file values have priority over the global file.

The IP Phone supports SSL and allows assignment of a username and password to authorize access to the HTTPS files.

In DHCP mode, the values from the DHCP server have a higher priority than the values in the global XML configuration files and the phone-specific XML configuration files.

Table C-2 below shows the XML configuration parameters. Table C-3 shows selections available for the time zones. A field in the XML configuration file only if the field requires an assignment other than default. If a field is erased or commented-out, it will have no affect on the IP Phone memory; the phone will use the existing value in memory.

Deleting an unused field from the XML configuration file also reduces traffic between the provisioning server and the IP Phone during a boot cycle.

Table C-2: XML Configuration Parameters

Parameter	Description
<Network>	
Interface	Sets use of wired network or wireless network. To use the wired network, set this to “wired”. To use the wireless network using Wi-Fi dongle, set this to “wireless”.. The permitted value is wired or wireless The default is wired

Parameter	Description
IPMode	Sets use of IPv4, IPv6 address mode. “IPv4” uses only IPv4 address mode. “IPv6” uses only IPv6 address mode. And, “Both” uses both IPv4 address mode and IPv6 address mode. The permitted value is IPv4, IPv6 or Both The default is IPv4
Ipv4.NetMode	Sets use of DHCP or static addressing for IPv4. In static mode, IP parameters are set manually. In DHCP, the IP Phone requests addresses from the local DHCP server. The permitted value is dhcp or static. The default is dhcp
Ipv4.Ipaddress	The phone’s IP address for IPv4. The permitted value is {ip_address}. The default is 192.168.1.1
Ipv4.Subnet	The phone’s subnet-mask for IPv4.. The permitted value is {ip_address}. The default is 255.255.255.0
Ipv4.Gateway	The phone’s default gateway for IPv4.. The permitted value is {ip_address}. The default is 192.168.1.254
Ipv4.Dns1	IP address of the primary DNS server used to resolve names to IP addresses for IPv4.. The permitted value is {ip_address}. The default is 0.0.0.0
Ipv4.Dns2	IP address of the secondary DNS server used to resolve names to IP addresses for IPv4.. The permitted value is {ip_address}. The default is 0.0.0.0
Ipv4.DnsFromDhcp	Obtains the DNS server address through option number 6 from DHCP server for IPv4.. The permitted value is enable or disable. The default is enable.
Ipv6.NetMode	Sets use of DHCP or static addressing for IPv6. In static mode, IP parameters are set manually. In DHCP, the IP Phone requests addresses from the local DHCP server. The permitted value is dhcp or static. The default is dhcp
Ipv6.Ipaddress	The phone’s IP address for IPv6. The permitted value is {ip_address}. The default is 192.168.1.1
Ipv6.SubnetPrefixLen	The length of phone’s subnet-prefix for IPv6. The permitted value is 1 ~ 128. The default is 64
Ipv6.Gateway	The phone’s default gateway for IPv6. The permitted value is {ip_address}.

Parameter	Description
	The default is 192.168.1.254
Ipv6.Dns1	IP address of the primary DNS server used to resolve names to IP addresses for IPv6. The permitted value is {ip_address}. The default is 0.0.0.0
Ipv6.Dns2	IP address of the secondary DNS server used to resolve names to IP addresses for IPv6. The permitted value is {ip_address}. The default is 0.0.0.0
Ipv6.DnsFromDhcp	Obtains the DNS server address through option number 6 from DHCP server for IPv6. The permitted value is enable or disable. The default is enable.
Port.PC	Disable the network of PC port. The permitted value is autonego or disable. The default is autonego.
Dhcp.LeaseTime	Defines the lease time (second) of DHCP request. The permitted value is 600 to 4294967295. The default is 691200.
<QoS>	
Tos.Data	Sets Type of Service (ToS) value for Voice data packets. The permitted value is 0 to 255. The default is 0.
Tos.Sip	Sets Type of Service (ToS) value for SIP packets. The permitted value is 0 to 255. The default is 0.
Vlan.Phone.Active	Activates the VLAN function for the Phone port. The permitted value is enable or disable. The default is disable.
Vlan.Phone.Id	Assigns unique identifier of the VLAN for the LAN (voice) port The permitted value is 1 to 4094. The default is 1.
Vlan.Phone.Priority	Establishes the IEEE 802.1 priority for Ethernet frames at the voice port, including signaling and voice packets of the IP Phone. The permitted value is 1 to 7. The default is 5.
Vlan.PC.Active	Activate the VLAN function for the PC port. The permitted value is enable or disable. The default is disable.
Vlan.PC.Id	Assigns unique identifier of the VLAN for the PC port. The permitted value is 1 to 4094. The default is 1.
Vlan.PC.Priority	Establishes the IEEE 802.1 priority for Ethernet frames at the PC port of the IP Phone.

Parameter	Description
	The permitted value is 1 to 7. The default is 3.
Vlan.Confirm	Confirm the change of vlan. The permitted value is enable or disable. The default is disable.
Vlan.Policy	Select the way to set the voice vlan. The permitted value is manual or lldp or dhcp. The default is lldp.
<Dot1x>	
Active	Activates the supplicant operation of the IP Phone. The permitted value is enable or disable. The default is disable.
Id	Sets the supplicant username. The id length is 0 to 50. The default is blank.
Password	Sets the supplicant password. The password length is 0 to 50. The default is blank.
PassthruMode	Sets the option of 802.1x passthru mode. The permitted value is passthru_logoff or passthru or passthru_off. The default is passthru.
Trace	Activates the trace of the 802.1x. The permitted value is enable or disable. The default is disable.
<Lldp>	
Active	Activates the lldp operation of the IP Phone. The permitted value is enable or disable. The default is disable.
Timer	Set the transmission period (second) of lldp packet. The permitted value is 30 to 120. The default is 70.
Trace	Activates the trace of the LLDP. The permitted value is enable or disable. The default is disable.
<Vpn>	
Active	Activates the VPN operation of the IP Phone. The permitted value is enable or disable. The default is disable.
Profile	Set the Profile type of the VPN server. The permitted value is manual or upload. The default is manual.
Address	Sets the IP address or FQDN of the VPN server.

Parameter	Description
	The address length is 0 to 255. The default is 0.0.0.0
Port	Assigns the port number of the VPN server The permitted value is 1 to 65535. The default is 1194.
Id	Sets the authentication username for VPN connection. The id length is 0 to 50. The default is blank.
Password	Sets the authentication password for VPN connection. The password length is 0 to 50. The default is blank.
Hash	Specifies the hash function algorithm. The permitted value is sha1 or sha256. The default is sha1.
Protocol	Set the connection protocol with VPN server. The permitted value is udp or tcp. The default is udp.
UserOption	Defines the user option when connecting to VPN server. The option string length is 0 to 128. The default is blank.
Verb	Assigns the Verb of the VPN server The permitted value is 0 to 11. The default is 3.
<Nettime>	
SntpServer.Address	Sets the IP address or FQDN of the SNTP server from which the phone obtains time data. The address length is 0 to 60. The default is blank.
SntpServer.FromDhcp	Obtains the SNTP server address through option number 42 from DHCP server. The permitted value is enable or disable. The default is enable.
Interval	Sets interval, in hours, between time sync with SNTP server. The permitted value is 0 to 120. The default is 1.
TimeOffset.Timezone	Sets the time zone for the phone. The default is GMT+00:00.
TimeOffset.FromDhcp	Obtains the time offset through option number 2 from DHCP server. The permitted value is enable or disable. The default is enable.
TimeFormat	Sets the LCD display format for time. The permitted value is off or 12 or 24. The default is 12.
DateFormat	Sets the LCD display format for date. The permitted value is {date_format}.

Parameter	Description
	YYYY: 4 digit year, MM: 2 digit month, DD: 2 digit day, m: abbr. month name, /, - : special characters, YY : 2 digit year, M : 1~2 digit month, D : 1~2 digit day, d : abbr. day of week. The default is MM/DD.
Dst.Active	Activates automatic daylight savings time (DST) adjustment by the phone. The permitted value is enable or disable. The default is disable.
Dst.StartMonth	Sets month DST starts. The permitted value is 1 to 12, with 1 being January and 12 being December. The default is 3.
Dst.StartWeekOfMonth	Sets week of month DST begins. The permitted value is 1 to 7. One (1) is the first week; each subsequent number is the subsequent week. Seven (7) is the last week of the month regardless of which week the last week is. The default is 1.
Dst.StartDayOfWeek	Sets day of the week DST begins. The permitted value is 1 to 7(1 is Sunday and 7 is Saturday). The default is 2.
Dst.StartTime	Sets the hour of the day DST begins. The permitted value is 0 to 23. The default is 2.
Dst.StopMonth	Sets month DST ends. The permitted value is 1 to 12 with 1 being January and 12 being December. The default is 11.
Dst.StopWeekOfMonth	Sets week of month DST ends. The permitted value is 1 to 7, with 1 being the first week, each number thereafter being subsequent weeks. Seven (7) represents the last week in the month The default is 1.
Dst.StopDayOfWeek	Sets day of the week DST ends. The permitted value is 1 to 7, with 1 being Sunday and 7 being Saturday. The default is 1.
Dst.StopTime	Sets the hour of the day DST ends. The permitted value is 0 to 23. The default is 2.
<Voip>	
Server.Active	Enables use of proxy server for SIP requests. The permitted value is enable or disable. The default is enable.

Parameter	Description
Server.DeRegister	Enables de-registration of each line during a reboot prior to re-registering. The permitted value is enable or disable. The default is enable.
Server.OutboundProxyServer	When defined, sends all SIP requests to the outbound proxy server. The media stream is not routed through the outbound proxy. The server address length is 0 to 50. The default is blank.
Server.OutboundProxyPort	Defines the port number of the outbound proxy server. The permitted value is 1024 to 65535. The default is 5060.
Server.BackupProxyServer	Defines a backup proxy server in case VoIP line proxy server fails. The server address length is 0 to 50. The default is blank.
Server.BackupProxyPort	Defines the port number of the backup proxy server. The permitted value is 1024 to 65535. The default is 5060.
Server.ExtProxyServer	Defines the server address for geographic redundancy. The server address length is 0 to 50. The default is blank. In <ExtProxyServer> tag, id attribute is used, and the range is 1 to 8.
Server.ExtProxyPort	Defines the port number of servers for geographic redundancy. The permitted value is 1024 to 65535. The default is 5060. In <ExtProxyPort> tag, id attribute is used, and the range is 1 to 8.
Server.StunServer	Sets the STUN server IP address or FQDN The server address length is 0 to 50. The default is blank.
Server.StunServerPort	Assigns the port number of the STUN server The permitted value is 1024 to 65535. The default is 3478.
Server.Domain	Specifies the SIP service domain. The domain length is 0 to 50. The default is blank.
Server.UserAgent	Defines the value of user-agent header. The user agent length is 0 to 100. The default is blank.
TransportMode	Sets the transport mode of SIP messages The permitted value is udp or tcp or tls. The default is udp.
Srtp.Active	Enables the srtp function. The permitted value is enable or disable. The default is disable.
Srtp.Crypto1	Defines the srtp crypto. The permitted value is AES_CM_128_HMAC_SHA1_80 or

Parameter	Description
	ARIA_CM_128_HMAC_SHA1_80 or ARIA_CM_192_HMAC_SHA1_80. The default is AES_CM_128_HMAC_SHA1_80.
Srtp.Crypto2	Defines the srtp crypto. The permitted value is none, AES_CM_128_HMAC_SHA1_80 or ARIA_CM_128_HMAC_SHA1_80 or ARIA_CM_192_HMAC_SHA1_80. The default is ARIA_CM_128_HMAC_SHA1_80.
Srtp.Crypto3	Defines the srtp crypto. The permitted value is none, AES_CM_128_HMAC_SHA1_80 or ARIA_CM_128_HMAC_SHA1_80 or ARIA_CM_192_HMAC_SHA1_80. The default is none.
Srtp.SrtpOnly	Defines the only srtp session description in SDP. The permitted value is enable or disable. The default is disable.
Tls.Crypto1	Defines the tls crypto. The permitted value is AES_128_CBC_SHA or ARIA_128_CBC_SHA or AES_128_CBC_SHA256 or AES_256_CBC_SHA256. The default is AES_128_CBC_SHA.
Tls.Crypto2	Defines the tls crypto. The permitted value is none, AES_128_CBC_SHA or ARIA_128_CBC_SHA or AES_128_CBC_SHA256 or AES_256_CBC_SHA256. The default is ARIA_128_CBC_SHA.
Tls.Version	Defines the tls version. The permitted value is TLSv1.0 or TLSv1.2. The default is TLSv1.0.
Tls.Ecc	Enables the tls ecc mode. The permitted value is enable or disable. The default is disable.
LocalPort.RtpStart	Defines the RTP port for media packets. The permitted value is 23000 to 24000. The default is 23000.
LocalPort.Udp	Defines local UDP port used for SIP messages. The permitted value is 1024 to 65535. The default is 5060.
LocalPort.Tcp	Defines local TCP port used for SIP messages. The permitted value is 1024 to 65535. The default is 5060.
LocalPort.Tls	Defines local TLS port used for SIP messages. The permitted value is 1024 to 65535. The default is 5061.
Codec.Priority1	Defines the order of priority of codecs for negotiation.

Parameter	Description
	The permitted value is pcmu or pcma or g729 or g722_64 or ilbc or ilbc_13 or opus_nb or opus_wb. The default is pcmu.
Codec.Priority2	Defines the order of priority of codecs for negotiation. The permitted value is none or pcmu or pcma or g729 or g722_64 or ilbc or ilbc_13 or opus_nb or opus_wb. The default is pcma.
Codec.Priority3	Defines the order of priority of codecs for negotiation. The permitted value is none or pcmu or pcma or g729 or g722_64 or ilbc or ilbc_13 or opus_nb or opus_wb. The default is g729.
Codec.Priority4	Defines the order of priority of codecs for negotiation. The permitted value is none or pcmu or pcma or g729 or g722_64 or ilbc or ilbc_13 or opus_nb or opus_wb. The default is g722_64.
Codec.Priority5	Defines the order of priority of codecs for negotiation. The permitted value is none or pcmu or pcma or g729 or g722_64 or ilbc or ilbc_13 or opus_nb or opus_wb. The default is none.
Timer.RegisterExpires	Defines time, in seconds, that a registration expires, and re-registration is required. The Expire header in the registration message includes this value. The permitted value is 0 to 86400. The default is 3600.
Timer.SubscribeExpires	Defines time, in seconds, that a subscription expires. The permitted value is 0 to 86400. The default is 3600.
Timer.T1	Sets the lowest value, in milliseconds, for the retransmission of SIP messages. The permitted value is 100 to 640. The default is 500.
Timer.T2	Sets the highest value, in milliseconds, for the retransmission of SIP messages. The permitted value is 641 to 6400. The default is 4000.
Timer.T4	Defines the time (milliseconds) that a phone waits after receiving an ACK message and before terminating the transaction. The permitted value is 4000 to 32000. The default is 5000.
Timer.InviteRetrans	Defines the retransmission number in case of no response for INVITE message. The permitted value is 3 to 7. The default is 5.
Timer.GeneralRetrans	Defines the retransmission number in case of no response for other message except INVITE message.

Parameter	Description
	The permitted value is 3 to 11. The default is 5.
Timer.GeneralRequestTimeout	After sending other message except INVITE message, a phone waits for a final response until the retransmission number described in Timer. GeneralRetrans or T1*64 time. But, because T1*64 is too long, this time (second) is used. The permitted value is 3 to 32. The default is 8.
Timer.RegiRenewal	Defines the value to send REGISTER message after some percentage time of the Expire time value in 200 OK message for REGISTER message. 1(10%), 2(20%), ~, 9(90%) The permitted value is 1 to 9. The default is 5.
Timer.RegiRetry	Defines the number to do re-registration after this number of registration trial. The permitted value is 1 to 10. The default is 4.
Timer.RegiTimeoutRetryTimer	Defines re-registration timer (second) after registration retry number is ended. The permitted value is 30 to 600. The default is 180.
Timer.RegiTimeoutRetryTimer1	Defines re-registration timer (second) after the time of registration retry number is ended. The permitted value is 1 to 100. The default is 10.
Options.OptionsTimer	Defines the retransmission timer (second) for option message. The permitted value is 0 to 6400. The default is 0.
Feature.MsgUrl	Defines the voice mail address. The url length is 0 to 50. The default is blank.
Feature.Blf	Specifies use of the BLF for notifications of DSS line status. The permitted value is enable or disable. The default is enable.
Feature.PrackMode	Enables the use of ACK response to provide reliable response to SIP requests. The permitted value is enable or disable. The default is disable.
Feature.UserParam	Defines “user=phone” parameter in the To, From, and Contact headers of REGISTER messages. The permitted value is enable or disable. The default is disable.
Feature.ReplacesMode	Enables use of the SIP “Replace” header. The permitted value is enable or disable.

Parameter	Description
	The default is enable.
Feature.RingNoanswerTime	Sets the timer value (second) of no answer for an incoming call. The permitted value is 0 to 3600. The default is 300.
Feature.AlertNoanswerTime	Sets the timer value (second) of alerting for an outgoing call. The permitted value is 0 to 3600. The default is 180.
Feature.SubscribeRetryTime	Sets the transmission period (second) of Subscribe message. The permitted value is 1 to 3600. The default is 60.
Feature.RefuseCause	Defines the cause value to refuse an incoming call. The permitted value is 400 to 699. The default is 603.
Feature.PhoneContext	Enables the phone-conext parameter in SIP message. The phone context length is 0 to 64. The default is blank.
Feature.CheckDomain	Enables the check of host address for an incoming call. The permitted value is enable or disable. The default is enable.
Feature.CheckTo	Enables the check of to header for an incoming call. The permitted value is enable or disable. The default is enable.
Feature.HoldSdpIp	In the hold state, uses phone IP instead of “0.0.0.0” in connection IP of SDP. The permitted value is enable or disable. The default is enable.
Feature.MultiIpSync	Enables geographic redundancy feature The permitted value is enable or disable. The default is disable.
Feature.PaiDisp	Uses the PAI header The permitted value is enable or disable. The default is enable.
Feature.PtimeInSdp	Enables to Include ptime in SDP message The permitted value is enable or disable. The default is disable.
Feature.Qvalue	Sets the Q value in the registration message. The permitted value is enable or disable. The default is disable.
Feature.Rport	Sets the rport parameter in the via header. The permitted value is enable or disable. The default is disable.
Feature.SimpleBlf	Assigns Blf feature in programmable buttons The permitted value is enable or disable. The default is enable.

Parameter	Description
Feature.SimpleBla	Assigns Bla feature in programmable buttons The permitted value is enable or disable. The default is enable.
Feature.ViaBranch	Enables the check for branch id in the via header. The permitted value is enable or disable. The default is disable.
Feature.RtpOnHeld	Enables to send RTP message in held state. The permitted value is enable or disable. The default is disable.
MaxLineNum	Sets the number of flexible buttons allocated for SIP lines. The permitted value is 1 to 72. The default is 1.
<VoipLine>	
Line.Server.Active	The permitted value is enable or disable. The default is enable. In <Line > tag, id attribute is used, and the range is 1 to 72.
Line.Server.ProxyAddress	Each line requires the address of the SIP proxy server. The address length is 0 to 50. The default is blank.
Line.Server.ProxyPort	Assigns the port number for the SIP proxy server. The permitted value is 0 to 65535. The default is 5060.
Line.Server.Displayname	Specifies the caller identification for the line, the phone sends the caller id for the line. The display name length is 0 to 50. The default is blank.
Line.Server.Name	Defines the SIP user id that is often a number for use when registering. Enter a number without dashes, enter 555-0100 as 5550100. The name length is 0 to 50. The default is blank.
Line.Server.Authname	Specifies the name for authentication if the proxy server challenges the line registration. It is required only if a proxy server requires authentication. The authentication name length is 0 to 50. The default is blank.
Line.Server.Password	Specifies the password for authentication if the proxy server challenges the line registration. The password length is 0 to 50. The default is blank.
Line.Type	Defines the type of line. The permitted value is as follows: ・ **private:** Calls to the line are presented to all appearances of the line but, an active call is visible and accessible through the original endpoints involved in setting up the call.

Parameter	Description
	• **shared (BLA):** Calls to the line are presented to all call appearances of the line and are visible and accessible by all authorized endpoints. • **dss:** The users can call to the covered extensions (Direct Station Selection) and answer incoming calls. The extension number must set in the Line.Extension field. The default is private.
Line.Extension	The associated extension number, which must be assigned for DSS line types only. The extension length is 0 to 50. The default is blank
Line.Ring.Type	Specifies the ring type for the line. The permitted value is default or off or 1 to 999. The default is default.
Line.Ring.DelayedTimer	Specifies a delay in the audible ring for the line. Delayed ring is adjustable from 0 to 10 seconds in 1-second increments The permitted value is 0 to 10. The default is 0.
Line.Feature.AutoAnswer	Enables auto answer feature for each line. The permitted value is enable or disable. The default is disable.
Line.Feature.DialTone	Specified the number of Dialtone file. The permitted value is none, 0 to 3. The default is 1.
Line.Feature.MsgUrl	Defines the voice mail address, IP address or phone number format for each line. The url length is 0 to 50. The default is blank.
Line.Feature.Dn	Defines the DN of BLA line when the main line is BLA. The dn length is 0 to 50. The default is blank.
Line.Feature.QValue	Defines the Q value for each line. The qvalue string length is 0 to 50. The default is blank.
<Dsp>	
VolRx.Handsfree	Specifies the rx gain level for the phone's handsfree speaker. The permitted value is 1 to 11. The default is 7.
VolRx.Handset	Specifies the rx gain level for the phone's handset. The permitted value is 1 to 11. The default is 6.
VolRx.MaxHandset	Defines the maximum level of rx gain for the phone's handset. The permitted value is 1 to 11. The default is 11.
VolRx.Headset	Specifies the rx gain level for the wired headset connected to the phone.

Parameter	Description
	The permitted value is 1 to 11. The default is 7.
VolRx.MaxHeadset	Defines the maximum level of rx gain for the phone's headset. The permitted value is 1 to 11. The default is 11.
VolTx.Handsfree	Specifies the tx gain level for the phone's handsfree speaker. The permitted value is 1 to 11. The default is 8.
VolTx.Handset	Specifies the tx gain level for the phone's handset. The permitted value is 1 to 11. The default is 8.
VolTx.Headset	Specifies the tx gain level for the wired headset connected to the phone. The permitted value is 1 to 11. The default is 8.
KeyTone.Active	Controls the Key tone operation. If enabled, when a button on the IP Phone is pressed the key tone is heard. The permitted value is enable or disable. The default is enable.
DefaultTonePath	The permitted value is handset or headset or speaker. The default is speaker.
Ring.Active	Toggles the ringer on and off. The permitted value is enable or disable. The default is enable.
Ring.Volume	Specifies the gain level for the phone's ring. The permitted value is 1 to 11. The default is 6.
Ring.Type	Specifies the default ring type for the IP Phone. Ring type can be a value from 1 to 4, 101 to 108 and 201 to 205: • 1 ~ 4, ring type 1 to 4 (tone ring) • 101 ~ 108, ring type 101 to 108 (wav file ring) • 201 ~ 205, ring type 201 to 205 (rtttl ring) The default is 1.
Ring.AertType.External	Specifies the ring type of external calls for the IP Phone. The permitted value is same as Ring.Type. The default is default.
Ring.Rtttl	Defines an RTTTL (Ring Tone Text Transfer Language) format text as ring type. The rtttl length is 0 to 512. The default is blank. In <Rtttl > tag, id attribute is used, and the range is 1 to 5.
Dtmf.Type	Sets the method to determine RFC 2833 payload type. The permitted value is: • inband — Always generate in-band DTMF. • nego — Negotiate RFC2833 with remote side. • rfc2833 — Always generate DTMF digits out-of-band RFC2833. • info — Always generate DTMF digits out-of-band in INFO message. The default is info.

Parameter	Description
Dtmf.Payload	Sets the DTMF payload type for RFC2833. The permitted value is 96 to 255. The default is 101.
Dtmf.Duration	Sets the DTMF duration. The permitted value is 50 to 300. The default is 120.
Dtmf.VolRfc2833	Sets the value of RFC-2833 rx volume. The permitted value is 1 to 10. The default is 13.
Dtmf.LongDtmf	Enables to play dtmf tone while pressing a digit. The permitted value is enable or disable. The default is disable.
Dtmf.LocalDtmfTone	Enables the local dtmf tone play. The permitted value is enable or disable. The default is disable.
Vad	Sets voice activation detection (VAD). The permitted value is enable or disable. The default is disable.
FrameDuration	Sets the RTP frame duration in milliseconds. The permitted value is 10 to 100. The default is 20.
Equalizer	Specifies the handset voice call equalizer. The permitted value is std, high and low. The default is std.
<Dial>	
Timer.FirstDigit	Defines the timer (second) to wait a first digit. The permitted value is 10 to 120. The default is 60.
Timer.Pause	Specifies the pause timer, in seconds, for the dial plan. The permitted value is 1 to 30. The default is 3.
Timer.ReleaseWait	Defines the play time (second) of alerting tone when a call is disconnected in 2nd call or conference call because of an error. The permitted value is 0 to 300. The default is 5.
Timer.PauseDial	Sets the timer value(millisecond) for pause dial The permitted value is 100 to 10000. The default is 500.
LockOutgoingCall	Unlocks the phone so a password is not required to place an outgoing call. The permitted value is enable or disable. The default is disable.
SpeedNumber	Assigns use of 3-digit Speed number dialing. The permitted value is enable or disable. The default is disable.

Parameter	Description
EndOfDigit	Defines the key to use as the end of digit. The permitted value is none or * or #. The default is #.
EndOfDigitSkip	If EndOfDigit is first digit, EndOfDigit is not processed. for example, when the value of EndOfDigitSkip is 1 and a user press ##, # is processed as an outgoing call. The permitted value is 1 to 10. The default is 1.
Digitmap.Normal	Defines a digit map string for the dial plan. The digitmap normal length is 0 to 1000. The default is blank.
Digitmap.Dialtone	Defines the digit map string to deliver second dial tone for a dial plan. The digitmap dialtone length is 0 to 1000. The default is blank.
Digitmap.Emergency	Defines the emergency call number. The digitmap emergency length is 0 to 1000. The default is blank.
Digitmap ProcessNumber	Defines the digit map process for a dial plan. The digitmap process number length is 0 to 1000. The default is blank. In <Process > tag, id attribute is used, and the range is 1 to 6.
Digitmap ProcessTrunc	Defines the number of digits to strip The permitted value is 0 to 30. The default is 0.
Digitmap ProcessPrefix	Defines the digits to insert in front of the number The prefix length is 0 to 32. The default is blank.
Digitmap ProcessPostfix	Defines the digits to insert at the end of the number. The postfix length is 0 to 32. The default is blank.
<Call>	
Limit.PerLine	Defines the number of active calls on a line. The permitted value is 1 to 10. The default is 10.
Limit.Total	Defines the total number of calls in a phone. The permitted value is 1 to 100. The default is 35.
BlockCallerId	Controls caller ID blocking. When enabled, the phone sends the anonymous caller id. The permitted value is enable or disable. The default is disable.
BlockAnonymousCall	Controls blocking incoming calls with an anonymous caller id. The permitted value is enable or disable. The default is disable.

Parameter	Description
HoldNoti.Active	Enables to ring the phone if the user returns to idle with a call on hold. The permitted value is enable or disable. The default is enable.
HoldNoti.OneTimeTone	Uses one time play of hold notification tone. The permitted value is enable or disable. The default is enable.
MsgWaitingTone	Enables message notification sound The permitted value is enable or disable. The default is enable.
DialTone	Uses a dial tone file. The permitted value is enable or disable. The default is disable.
CallWaiting.Active	Controls visual notification of incoming call while busy. The permitted value is enable or disable. The default is enable.
CallWaiting.NotiTone	Controls audible notification when Call Waiting is enabled The permitted value is enable or disable. The default is enable.
CallWaiting.OneTimeTone	Uses one time play of call waiting tone. The permitted value is enable or disable. The default is disable.
AutoAnswer.Active	Controls answering an incoming call automatically activating the speakerphone or headset. The permitted value is enable or disable. The default is disable.
AutoAnswer.NotiTone	Uses the notification tone play when answering automatically. The permitted value is enable or disable. The default is enable.
DirectCall.Active	Controls Hotline/Warm-Line operation. Going off-hook automatically places a call to the designated party. The permitted value is enable or disable. The default is disable.
DirectCall.Number	Defines the destination for a direct call. The number length is 0 to 50. The default is blank.
DirectCall.Timer	Sets the delay (second) for Warm Line operation. After the timer expires, the direct call is made. Set to zero (0) for Hotline. The permitted value is 0 to 30. The default is 5.
ConfAllCallRel	Enables to release all calls when a host releases a conference call. The permitted value is enable or disable. The default is disable.
Deflect	Enables to transfer a call when a user presses CallFwd softkey in an incoming call. The permitted value is enable or disable. The default is enable.

Parameter	Description
Dnd	Disables dnd function for the phone. The permitted value is enable or disable. The default is enable.
FeatureSync	Enables to synchronize with a call server for the call forward and dnd feature. The permitted value is enable or disable. The default is enable.
Ignore180After183	Enables to ignore 180 messages after receiving 183 messages. The permitted value is enable or disable. The default is enable.
IgnoreMissedCallInDial	Enables to display a missed call, in the dialing state, when answering an incoming call. This feature is operated when call waiting feature is disabled. The permitted value is enable or disable. The default is enable.
RecordByPhone	Enables the call recoding feature by a phone. The permitted value is enable or disable. The default is disable.
SharpConvert	Enables to convert dialed digit '#' to '%23'. The permitted value is enable or disable. The default is disable.
SplitConf	Enables to split a local conference call. The permitted value is enable or disable. The default is disable.
SystemConferenceOnly	Enables system conference feature not using a local conference by a phone. The permitted value is enable or disable. The default is disable.
VoiceRecord	Enables voice recording feature. The permitted value is enable or disable. The default is enable.
BlindTransfer	Enables the blind transfer feature when pressing DSS/BLF button or Speed dial during a call. If disabled, transfer feature is operated. The permitted value is enable or disable. The default is disable.
OverlapDial.Active	Enables the overalp dial mode. The permitted value is enable or disable. The default is enable.
OverlapDial.Code	Defines the code value to enable overlap dial mode. The code length is 0 to 10. The default is blank.
HoldDisconnect	Enables to terminate the call without holding the original call by pressing BLA line with the remote hold during a call. The permitted value is enable or disable. The default is disable.

Parameter	Description
AlternateLineOnBlaBusy	Defines that the next step is selected as one of bla, private and none, if the main line is BLA, when selecting the line, the line is busy. The permitted value is bla, private, none. The default is private.
TransferBefore180	Enables to send REFER msg., when trying to transfer, if the transferor terminates the call before receiving 180 msg. The permitted value is enable or disable. The default is disable.
<Directory>	
Xml.Active	Activates the external (Remote) Phone Book function. The permitted value is enable or disable. The default is disable.
Xml.Url	Defines the http server url for the external (Remote) Phone Book. The url length is 0 to 255. The default is blank.
Xml.Id	Sets an authentication username for XML server connection. The id length is 0 to 50. The default is blank.
Xml.Password	Sets an authentication password for XML server connection. The password length is 0 to 50. The default is blank.
<Pui>	
PhoneName	Assigns a name to the phone for idle mode display. The phone name is 0 to 24. The default is blank.
Password.Admin	Assigns admin password for LCD Configuration Menu and Phone Unlock. The password length is 0 to 50. The default is blank.
Password.User	Assigns user password for LCD Configuration Menu and Phone Unlock. The password length is 0 to 50. The default is blank.
HeadsetMode	Toggles the headset mode on and off. The permitted value is enable or disable. The default is disable.
LcdContrast	Specifies the contrast level for the phone's LCD in 1010i and 1020i. The permitted value is 1 to 10. The default is 6.
LcdDimming	Enables LCD dimming feature in 1020i. The permitted value is enable or disable. The default is disable.
LcdBright	Specifies the brightness level for the phone's LCD in 1030i, 1040i and 1050i. The permitted value is 1 to 10. The default is 6.

Parameter	Description
Backlight.Active	Enables LCD Backlight feature in 1020i, 1030i, 1040i and 1050i. The permitted value is enable or disable. The default is enable.
Backlight.Timer	Defines the timer value(second) to turn on LCD backlight in 1020i, 1030i, 1040i and 1050i. The permitted value is off or 5 or 10 or 20 or 30 or 60 or 180 or forever. The default is 5.
Display.IdleScreen	Sets the LCD screen display is line button mode or calendar mode in idle screen in 1030i, 1040i and 1050i. The permitted value is default or calendar. The default is default.
Display.BackgroundImage	Sets the background image of LCD screen in 1030i, 1040i and 1050i.
Display.LabelTransparency	Defines the value of Label Transparency in 1030i, 1040i and 1050i. The permitted value is 0 to 100. The default is 100.
Display.LabelMaxWidth	Defines the maximum width value of Label in 1030i, 1040i and 1050i. The permitted value is 0, 50 to 202. The default is 202.
Display.ScreenTimeout	Defines the screen timeout value to turn off backlight when there is no user action in 1030i, 1040i and 1050i. The permitted value is 5, 10, 15, 30, 60 or 240 minutes. The default is 10 minutes.
Display.ScreenTimeoutCall	Defines the screen timeout value to turn off backlight when there is no user action during a call in 1030i, 1040i and 1050i. The permitted value is 5 to 600. The default is 240.
Display.ScreenSaver	Defines the type of screen saver in 1030i, 1040i and 1050i. The permitted value is Blank or Clock. The default is Clock.
Display.DisplayDtmf	Enables to display the digit in the LCD screen when pressing a digit in a call state. The permitted value is enable or disable. The default is disable.
Display.DisplayNameOnCmdCall	Enables to display the display name not number in Command Call. The permitted value is enable or disable. The default is disable.
Display.BlaMenu	Enables to display BLA setting in the programmable button setting menu of LCD menu. The permitted value is enable or disable The default is disable.
Display.DiversionInfo	Enables to display the number of original call when picking up a call. The permitted value is enable or disable. The default is disable.
Display.SrtpTitle	Enables to display SRTP notification title not icon.

Parameter	Description
	The permitted value is enable or disable. The default is disable.
Display.DisplayLineClog	Enables to display line information in Call log. The permitted value is enable or disable. The default is disable.
Display.LinePageTimer	Defines the timer value (second) to move 1st Page from other page in 1020i, 1030i, 1040i and 1050i. The permitted value is off or 15 or 30 or 60. The default is off.
Display.LinePages	Defines the number of Page in 1020i, 1030i, 1040i and 1050i. The permitted value is 1 to 3. The default is 3.
Display.PbHigherPriority	Enables to display the name in phonebook in an incoming call. The permitted value is enable or disable. The default is disable.
Display.UpDownPreview	Enables to show all calls with Up or Down navigation key. The permitted value is enable or disable. The default is disable.
Display.MissedCallNoti	Enables to display missed call notification. The permitted value is enable or disable. The default is enable.
Display.CalendarBgImg	Enables the background image of Calendar mode on 1030i, 1040i and 1050i. The permitted value is enable or disable. The default is enable.
Display.SimpleCallInfoTimer	Defines the value of timer to display the simple call information in the Top-bar during a call. The permitted value is off, immediately or 1- 999. The default is off.
Display.Dial	Defines the screen display type when dialing. The permitted value is default or button. The default is default.
Call.DirectDial	Enables to dial the selected number by off-hook in phonebook and call log. The permitted value is enable or disable. The default is disable.
Call.GroupListen	Enables to provide incoming audio over the speaker with outgoing audio from the handset. The permitted value is enable or disable. The default is disable.
Call.HoldNewcall	Enables to create automatically a new call and go off-hook when requesting Hold. The permitted value is enable or disable. The default is disable.

Parameter	Description
Call.UnregisteredOffhook	Enables not to go off-hook in registration failure state. The permitted value is enable or disable. The default is disable.
Call.OnetouchPickup	Enables to go the call state not hold state by pressing a line key when there are over 2 calls. The permitted value is enable or disable. The default is enable.
Call.OnhookByLine	Enables to go on-hook by pressing a line key. The permitted value is enable or disable. The default is disable.
Call.Predialing	Enables to dial after collecting numbers. The permitted value is enable or disable. The default is disable.
Call.Prespeaker	When changing the sound path form handset to speaker mode, after handset goes on-hook, speaker mode is enabled. The permitted value is enable or disable. The default is disable.
Call.RestoreCallOnTransferFail	Enables to return automatically to the previous call when transfer is failed. The permitted value is enable or disable. The default is disable.
ReturnActCall.Timer	Defines the timer value (second) to return active call. The permitted value is 0 to 999. The default is 5.
RingLed.OnColor	Defines the RGB color code value in on-time of Ring LED in 1040i and 1050i. The default is #0000FF00.
RingLed.OffColor	Defines the RGB color code value in off-time of Ring LED in 1040i and 1050i. The default is #00000000.
Info.Admin	Defines the output permission of LCD menu in admin account when LCD menu lock is enabled. The permitted value is one of network, sip, server, dss, provision, version. It can be combined with semicolons. The default is network;sip;server;provision;version.
Info.User	Defines the output permission of LCD menu in user account when LCD menu lock is enabled. The permitted value is one of network, sip, server, dss, provision, version. It can be combined with semicolons. The default is provision;version.
Info.Anybody	Defines the output permission of LCD menu when LCD menu lock is disabled. The permitted value is one of network, sip, server, dss, provision, version. It can be combined with semicolons.

Parameter	Description
	The default is provision;version.
PrimeLinekey	Defines the button number to use the prime line. The permitted value is 1 to132. The default is 1.
ExtensionStart	Defines the start number for the button of the extension module. The permitted value is 0 to144. The default is 49.
Prog.Function	Specifies the function for programmable buttons. The permitted value is as follows: none speed phonebook calllog mute headset redial return forward conference transfer hold dnd ringoff message dssblf xmlservice voicerecord dssbla blindxfer sysconf voipline queuestatus featurekey In <Prog > tag, id attribute is used, and the range is 1 to 132.
Prog.Number	Specifies the number for dssblf or dssbla or speed. The number length is 0 to 60. The default is blank.
Prog.VoipLine	Specifies the number of voipline when <Function> is assigned to voipline. The permitted value is none or 1 to 72. The default is 1.
Prog.Extension	Specifies the automatic outgoing dialing number when <Function> is assigned to dssblf or dssbla. The extension length is 0 to 60. The default is blank.

Parameter	Description
Prog.DssBlf	Specifies the transfer mode when <Function> is assigned to dssblf. The permitted value is none or xfer or blind_xfer. The default is none.
Prog.LineFeature.Ring	Specifies the ring type for the line. The permitted value is default or off or 1 to 999. The default is default.
Prog.LineFeature.DelayedTimer	Specifies a delay in the audible ring for the line. Delayed ring is adjustable from 0 to 10 seconds in 1-second increments The permitted value is 0 to 10. The default is 0.
Prog.LineFeature.AutoAnswer	Enables auto answer feature for the line. The permitted value is enable or disable. The default is disable.
Prog.LineFeature.DialTone	Specifies the number of Dialtone file for the line. The permitted value is none, 0 to 3. The default is 1.
Prog.Url	Specifies the URL for the XML service to connect using HTTP/S. The URL length is 0 to 255. The default is blank.
Prog.Label	Specifies the label for programmable buttons. The label length is 0 to 24. The default is blank.
<System>	
Account.Password	Defines the password for “admin” account. <account> tag uses attribute id=”3”. The password length is 0 to 50.
Account.Password	Defines the password for “user” account. <account> tag uses attribute id=”4” The password length is 0 to 50.
Account.Id	Defines the username for a web admin user. <account> tag uses attribute id=”5” The id length is 0 to 24. The default is blank.
Account.Password	Defines the password for a web admin user. <account> tag uses attribute id=”5” The password length is 0 to 50. The default is blank.
Account.Id	Defines the username for a web admin user. <account> tag uses attribute id=”6” The id length is 0 to 24. The default is blank.
Account.Password	Defines the password for a web admin user. <account> tag uses attribute id=”6” The password length is 0 to 50. The default is blank.
Language	Specifies the language displayed in the LCD. The default is English_Default.

Parameter	Description
Utf8	The permitted value is enable or disable. The default is disable.
WebServer.Https.Active	Enables access to the Web Manager. The permitted value is enable or disable. The default is disable.
WebServer.Https.Port	The permitted value is 1 to 65535. The default is 443.
WebServer.Timer	Defines the timeout value (second) of web server connection. The permitted value is 0 to 3600. The default is 600.
WebServer.WhiteList	Defines access control lists for web server. The list length is 0 to 255. The default is blank.
SecureShell.Active	Enables access to the Secure Shell for debugging. The permitted value is enabled to disable. The default is disable.
SecureShell. WhiteList	Defines access control lists for secure shell. The list length is 0 to 255. The default is blank.
Llo.Active	Enables LLO (Line Lock Out) feature. The permitted value is enable or disable. The default is disable.
Llo.Timer	Defines the timer value (second) to start the LLO. The permitted value is 0 to 999. The default is 30.
XmlService.Active	Enables XML service. The permitted value is enable or disable. The default is enable.
XmlService.Url	Defines XML server url. The url length is 0 to 255. The default is blank.
XmlService.UrlDirectory	Defines the url for XML directory service. The directory url length is 0 to 255. The default is blank.
XmlService.UrlCalllog	Defines the url for XML call log service. The calllog url length is 0 to 255. The default is blank.
XmlService.Id	Sets the authentication username for XML server connection. The id length is 0 to 50. The default is blank.
XmlService.Password	Sets the authentication password for XML server connection. The password length is 0 to 50. The default is blank.

Parameter	Description
Sms.Active	Enables SMS feature. The permitted value is enable or disable. The default is enable.
Sms.KtMode	Enables KT mode for SMS service. The permitted value is enable or disable. The default is disable.
EasySetup	Enables easy setup feature. The permitted value is enable or disable. The default is enable.
Log.Trace	Sets the hexadecimal number combined every bit-fields by separating with semicolons for each module to gather the trace messages from each modules of the phone.
Syslog.Active	Enables syslog feature. The permitted value is enable or disable. The default is disable.
Syslog.Address	Defines syslog server address. The address length is 0 to 120. The default is blank.
Syslog.MaxSize	Defines the maximum size of syslog file. The MaxSize is 10 to 5120. The default is blank.
Syslog.Volatility	When it is enable, the phone will save the log to the volatile memory. The permitted value is enable or disable. The default is enable.
USB	Enables USB port in 1040i and 1050i. The permitted value is enable or disable. The default is enable.
<Provision>	
Type	Defines the protocol of the provisioning server. The permitted value is tftp or http. The default is tftp.
SwDownload	Enables download of firmware from the provisioning server. The permitted value is enable or disable. The default is enable.
CfgDownload	Enables download of configuration file from the provisioning server. The permitted value is enable or disable. The default is enable.
Tftp .Address	Defines the address of TFTP provisioning server. The address length is 0 to 120. The default is blank.
Tftp .NameFormat	Specifies the filename of phone specific configuration file. The name or path must include “$MAC”, which is replaced by the MAC address during provisioning.

Parameter	Description
	The name format length is 0 to 50. The default is $MAC.xml.
Tftp .FromDhcp	Obtains the TFTP server address through option number 66/150 from DHCP server. The permitted value is enable or disable. The default is enable.
Http.Url	Defines the HTTP/S provisioning server URL The url length is 0 to 255. The default is blank.
Http.Id	Specifies the ID for authenticating to an HTTPS session. The id length is 0 to 50. The default is blank.
Http.Password	Specifies the password for authenticating to an HTTPS session. The password length is 0 to 50. The default is blank.
Http.NameFormat	Specifies the filename of phone specific configuration file. The name or path must include “$MAC”, which is replaced by the MAC address during provisioning. The name format length is 0 to 50. The default is $MAC.xml.
Http.Verification	Defines HTTPS certification verification method. The permitted value is none or cert or all. The default is all.
Http.FromDhcp	Enables to obtain the HTTP/S server address through option number from DHCP server. The permitted value is enable or disable. The default is enable.
Http.DhcpOptionNum	Defines the DHCP Option for the HTTP/HTTPS server URL. The permitted value is 1 to 255. The default is 160.
SmartUpgrade.Active	Enables Smart Update, to check for and download firmware periodically. The permitted value is enable or disable. The default is disable.
SmartUpgrade.Download	Enables download of configuration file or firmware or both by smart upgrade. The permitted value is cfg or sw or both. The default is both.
SmartUpgrade.Type	Defines the timing of Smart Upgrade as an interval or time of day. The permitted value is interval or time. The default is time.
SmartUpgrade.Interval	Specifies the interval, in seconds, between firmware checks. The permitted value is 1 to 86400. The default is 3600.
SmartUpgrade .SpecifiedTime.StartHour	Specifies the start hour for daily firmware checks. The permitted value is 0 to 23 The default is 0.

Parameter	Description
SmartUpgrade .SpecifiedTime.StartMinute	Specifies the start minute for daily firmware checks. The permitted value is 0 to 59. The default is 0.
SmartUpgrade .SpecifiedTime.EndHour	Specifies the end hour for daily firmware checks. The permitted value is 0 to 23 The default is 1.
SmartUpgrade .SpecifiedTime.EndMinute	Specifies the end minute for daily firmware checks. The permitted value is 0 to 59. The default is 0.
SmartUpgrade.Confirm	Enables user confirmation of firmware download from the provisioning server. The permitted value is enable or disable. The default is enable.
CfgUpload .Active	Enables the configuration upload feature. The permitted value is enable or disable. The default is disable.
CfgUpload .Url	Defines the url to upload the configuration. The url length is 0 to 255. The default is blank.
CfgUpload .Id	Set the authentication username for configuration upload. The id length is 0 to 50. The default is blank.
CfgUpload .Password	Set the authentication password for configuration upload. The password length is 0 to 50. The default is blank.
Login.Active	Enables login feature. The permitted value is enable or disable. The default is disable.
Login.Persistent	Enables to stay logged in during reboots. Login screen doesn't display. The permitted value is enable or disable. The default is disable.
Login.Id	Defines a username to log in. It is used as a file name (Id.xml) to download a user profile. The id length is 0 to 50. The default is blank.
Login.Password	Defines a password to log in. It is used for accessing a user profile. The password length is 0 to 50. The default is blank.
Login.MaxPeriod	Defines Maximum permitted login time (Hour). This value is valid when the value of <Active> is enable. • • 0: permit to stay logged without logout in power on state. A user can select No timeout, 1 - 24., • 1 ~ 24(default): A user can select 1 – MaxPeriod. The default is 24.

Parameter	Description
<VQM >	
Vqm.SipRtcpXr.SessionReport	Enables Voice Quality Measurement feature. The permitted value is enable or disable. The default is disable.
Vqm.SipRtcpXr.CollectorAddress	Defines a collector address. The CollectorAddress length is 0 to 255. The default is blank.
Vqm.SipRtcpXr.CollectorPort	Defines a port of collector. The permitted value is 0 to 65535. The default is 5060.

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        </Vqm>
    </Settings>
```

Table C-3: Time Zones Chart

Country	Country
GMT-12:00 International Date Line, West	GMT+01:00 Oslo, Norway
GMT-11:00 Midway/Samoa	GMT+01:00 Paris, France
GMT-10:00 Hawaii	GMT+01:00 Rome, Italy
GMT-09:00 Alaska	GMT+01:00 Stockholm, Sweden
GMT-08:00 Los Angeles, California	GMT+01:00 Vienna, Austria
GMT-08:00 Pacific Standard	GMT+02:00 Athens, Greece
GMT-07:00 Mountain Standard	GMT+02:00 Harare, Zimbabwe
GMT-06:00 Central Standard	GMT+02:00 Helsinki, Finland
GMT-06:00 Mexico City, Mexico	GMT+02:00 Istanbul, Turkey
GMT-06:00 Saskatc., Canada	GMT+02:00 Jerusalem, Israel
GMT-05:00 Bogotá, Colombia	GMT+03:00 Kuwait
GMT-05:00 Eastern Standard	GMT+03:00 Moscow, Russia
GMT-05:00 Lima, Peru	GMT+03:00 Nairobi, Kenya
GMT-05:00 New York, NY	GMT+03:00 Riyadh, Saudi Arabia
GMT-04:00 Atlantic Standard	GMT+03:30 Tehran, Iran
GMT-04:00 Caracas, Venezuela	GMT+04:00 Abu Dhabi, UAE
GMT-04:00 La Paz, Bolivia	GMT+04:00 Tbilisi, Georgia
GMT-03:30 New Found land	GMT+04:30 Kabul, Afghanistan
GMT-03:00 Brasilia, Brazil	GMT+05:00 Islamabad, Pakistan
GMT-03:00 Buenos Aires, Argentina	GMT+05:30 New Delhi, India
GMT-03:00 Georgetown, Guyana	GMT+06:00 Dhaka, Bangladesh
GMT-02:00 Inuuk, Greenland	GMT+06:30 Yangon, Myanmar
GMT-01:00 Azores, Portugal	GMT+07:00 Bangkok, Thailand
GMT-01:00 Cape Verde	GMT+07:00 Jakarta, Indonesia
GMT 00:00 Casablanca, Morocco	GMT+08:00 Beijing, China
GMT 00:00 Lisbon, Portugal	GMT+09:00 Seoul, Korea
GMT 00:00 London, England	GMT+09:00 Tokyo, Japan
GMT+01:00 Amsterdam, Netherlands	GMT+09:30 Darwin, Australia
GMT+01:00 Berlin, Germany	GMT+10:00 Guam Standard
GMT+01:00 Bern, Switzerland	GMT+11:00 Solomon Islands
GMT+01:00 Brussels, Belgium	GMT+12:00 Marshall Islands
GMT+01:00 Central European	GMT+12:00 Auckland, Wellington
GMT+01:00 Copenhagen, Denmark	GMT+13:00 Nuku'alofa
GMT+01:00 Madrid, Spain	

Appendix D. Auto Provisioning Routine

General

The 1000i series phones incorporate software routines to automatically provision the configuration and software of the phone. To initialize LAN settings, DHCP to gather basic network and provisioning server information, downloads XML configuration and software files from the provisioning server, and registers and subscribes with the SIP call server using the configuration data.

The phone attempts contact with the provisioning server to download XML configuration files then checks for software updates. For more details on the XML configuration files, see Table C-2. Note downloads can be disabled separately for the configuration and software files. As a default, the phone will employ TFTP as the download protocol. Optionally, the phone can employ HTTP/HTTPS as the provisioning server protocol. If the DHCP server returns a URL for the HTTP/S Option (default Opt. #160), the IP Phone shifts provisioning protocol to HTTP/HTTPS using the URL provided to access the server.

DHCP

During the initial and subsequent boot-up cycles, the phone requests various data options from the DHCP server. The following lists the requested options.

- DHCP Option #1 Subnet Mask
- DHCP Option #2 Time offset in seconds of the local time zone from GMT.
- DHCP Option #3 Default Gateway (router) IP Address
- DHCP Option #6 DNS Server IP Address
- DHCP Option #42 SNTP Server IP Address
- DHCP Option #50 IP address of the phone
- DHCP Option #66 TFTP Server Name
- DHCP Option #150 TFTP Server IP Address has priority over Option 66 when set.
- DHCP Option #160 default HTTP/HTTPS URL for provisioning

Table D-1: Software Version file name

Model	Global XML configuration file name
1010i	VER_1010i_SIP
1020i	VER_1020i_SIP
1030i	VER_1030i_SIP
1040i	VER_1040i_SIP
1050i	VER_1050i_SIP

TFTP Downloads

When TFTP is employed, the IP Phone will contact the server provided from the DHCP server using the name or IP address from DHCP Option #66 or #150 or the TFTP server in memory when the DHCP option is not available. The IP Phone will request the TFTP server send the global XML configuration file. Data from the XML configuration file is transferred to the phone memory.

Changes in the provisioning data from the global XML configuration file are transferred to the phone memory and the download process is repeated for the phone-specific XML configuration file. Configuration data in the phone specific file overwrites data from the global file.

After completing download of the XML configuration files, the IP Phone requests the software version file in Table D-1 from the server. This file is an text file that identifies the software version available on the server as well as the names of software files to be downloaded.

The phone compares the software version identified in the file with the version in the phone. If different, the files identified in the version file are downloaded. Typically, the files required for download include the main firmware file and an xml file but may include other files as needed for the upgrade. These files along with the version file are available from Ericsson-LG Enterprise. If desired, comments may be added to the version file using “#” as the leading character of the comment line.

If software download is required, the IP Phone will enable the reboot flag and after completing the download cycle, the IP Phone reboots.

HTTP/HTTPS downloads

When HTTP/HTTPS is employed, the IP Phone will contact the URL defined in the memory or in the HTTPS DHCP Option, default #160. When the DHCP option is populated, the phone employs HTTP/HTTPS instead of the default TFTP mode. If the server uses HTTPS, the phone will establish an open SSL link. During the SSL handshake, the phone may optionally verify that the validity of the HTTPS Certificate of Authority (CA). The IP Phone verifies that the CA is from a valid authority such as Verisign®, the server providing the CA is correct and the certificate has not expired. The latter is verifiable only if an SNTP server provides time to the IP Phone. Each of these three verifications is enabled by default but can be disabled using the XML configuration file or manual configuration. The IP Phone supports several standard CAs including Verisign®. In addition, a CA may be copied to the phone using the Web Manager.

If the HTTP/HTTPS server requires authentication, the IP Phone uses the HTTPS ID and Password assigned; the IP Phone supports the basic and digest method. At this point, the phone will follow the same routine as for TFTP. The IP Phone requests the global XML configuration file. Data is transferred to memory.

Changes in the provisioning data from the global XML configuration file are transferred to the phone memory and the download process is repeated for the phone-specific file. Configuration data in the phone specific file overwrites data from the global file.

For HTTP/S, a different file name can be assigned in the phone to use as file name but the MAC address must be included as part of the file name or path.

After completing download of the configuration files, the IP Phone requests the software version file in Table D-1 from the server. This file is an text file that identifies the software version available on the server as well as the names of the files for download. The phone first requests the file “sw_mac address/VER_10X0i_SIP”, if not found, the file “VER_10X0i_SIP” is requested. The phone compares the software version identified in the file with the version in the phone. If different, the files identified in the version file are downloaded. Typically, the files required for download include the main firmware file but may include other files as needed for the upgrade. These files along with the version file are available from Ericsson-LG Enterprise.

If the software download is required, after completion of the download cycle the IP Phone reboots.

SIP Server Check-in Request

Upon receipt of a SIP check-in or resync request from the SIP server (Notify check sync event message), the IP Phone initiates the reboot process. The phone must be idle to initiate provisioning, if busy, upon return to idle, the process starts. During the process, the IP Phone downloads any XML configuration file changes as well as new software regardless of the settings for the software and configuration download flags.

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