IG6600-IP2061

Administration Manual

Environment

The phone you have purchased must not be disposed of with household waste. You should return these to your distributor if they are to replace or dispose of them in an approved recycling centre.

FCC Statement

This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions in this manual, may cause interference to radio communications. This equipment has been tested and found to comply with the limits for a Class B computing device pursuant to Subpart J of Part 15 of FCC rules, which are designed to provide reasonable protection against radio interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, in which case the user, at is own expense, will be required to take whatever measures are necessary to correct the interface.

CE Declaration of Conformity

This equipment complies with the requirements relating to electromagnetic compatibility, EN55022 class B for ITE and EN 50082-1. This meets the essential protection requirements of the European Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

Copyright Notice

Trademarks

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WARNING!

1. Read these installation instructions carefully before connecting the IP phone to its power adapter.
2. To reduce the risk of electric shock, do not remove the cover from the IP phone or attempt to dismantle it. Opening or removing covers may expose you to dangerous voltage levels. Equally, incorrect reassembly could cause electric shock on re-use of the appliance.
3. Do not expose the IP phone to fire, direct sunlight or excessive heat.
4. Do not expose the IP phone to rain or moisture and do not allow it to come into contact with water.
5. Do not install the IP phone in an environment likely to present a Threat of Impact.
6. You may clean the IP phone using a fine damp cloth. Never use solvents (such as trichloroethylene or acetone), which may damage the phone's plastic surface and LCD screen. Never spray the phone with any cleaning product whatsoever.
7. Take care not to scratch the LCD screen.
8. The IP phone is designed to work in temperatures from 0°C to 45°C (32°F to 104°F).
9. The IP phone must be installed at least 1 meter from radio frequency equipment, such as TVs, radios, hi-fi or video equipments (which radiate electromagnetic fields).
10. Do not connect the LAN/PC port to any network other than an Ethernet network.
11. Do not attempt to upgrade your IP phone in an unstable power environment. This could cause unexpected damages.
12. Do not work on the system during lightning storms. Please disconnect all cables.
13. Children don't recognize the risks of electrical appliances. Therefore use or keep the phone only under supervision of adults or out of the reach from children.
14. No repair can by performed by the end user, if you experience trouble with this equipment, for repair or warranty information, please contact your supplier.

Electrical Powering:

The IP2061 can be powered with PoE Switch or power adaptor, the power adaptor must be 5V/2A. Any damage caused to the IP2061 as a result of using unsupported power adaptors will not be covered by the manufacturer's warranty.

Product Disposal Warning:

Ultimate disposal of this product, accessories, packing, especially the batteries should be handled carefully for recycle and nature protection in accordance with national laws and regulations.

Table Of Contents

1. Getting Started

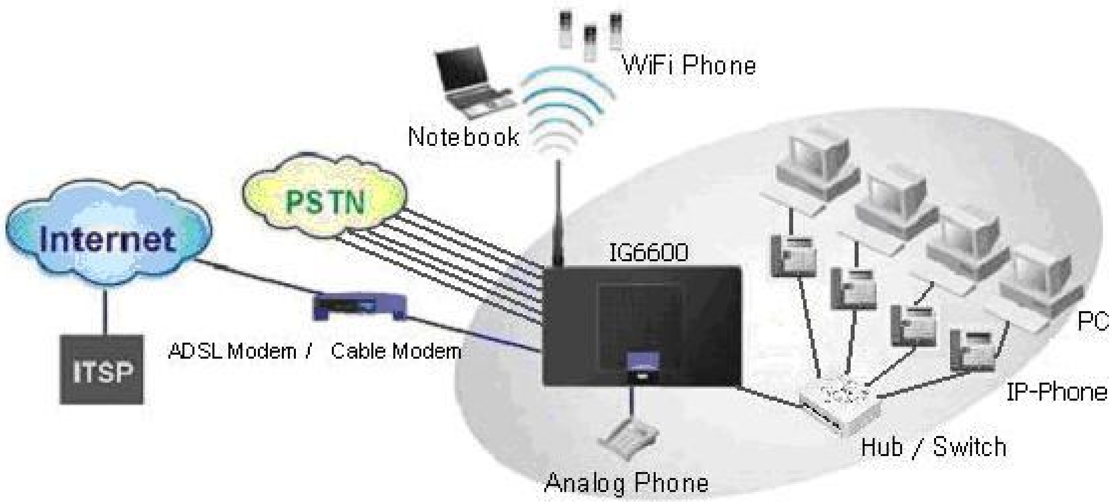
This section will help you quickly find the information that you need to make use of the full features of your IP2061.

2. Overview of the IP2061

The IP2061 is an internet telephony desktop phone that connects to intelligent gateway IG6600 with an Ethernet cable rather than traditional a PSTN line. Basically, it needs to be connected to LAN side or WAN side of IG6600. Like a traditional office telephone, it can deliver good voice quality and perform a great number of PBX-equivalent call features.

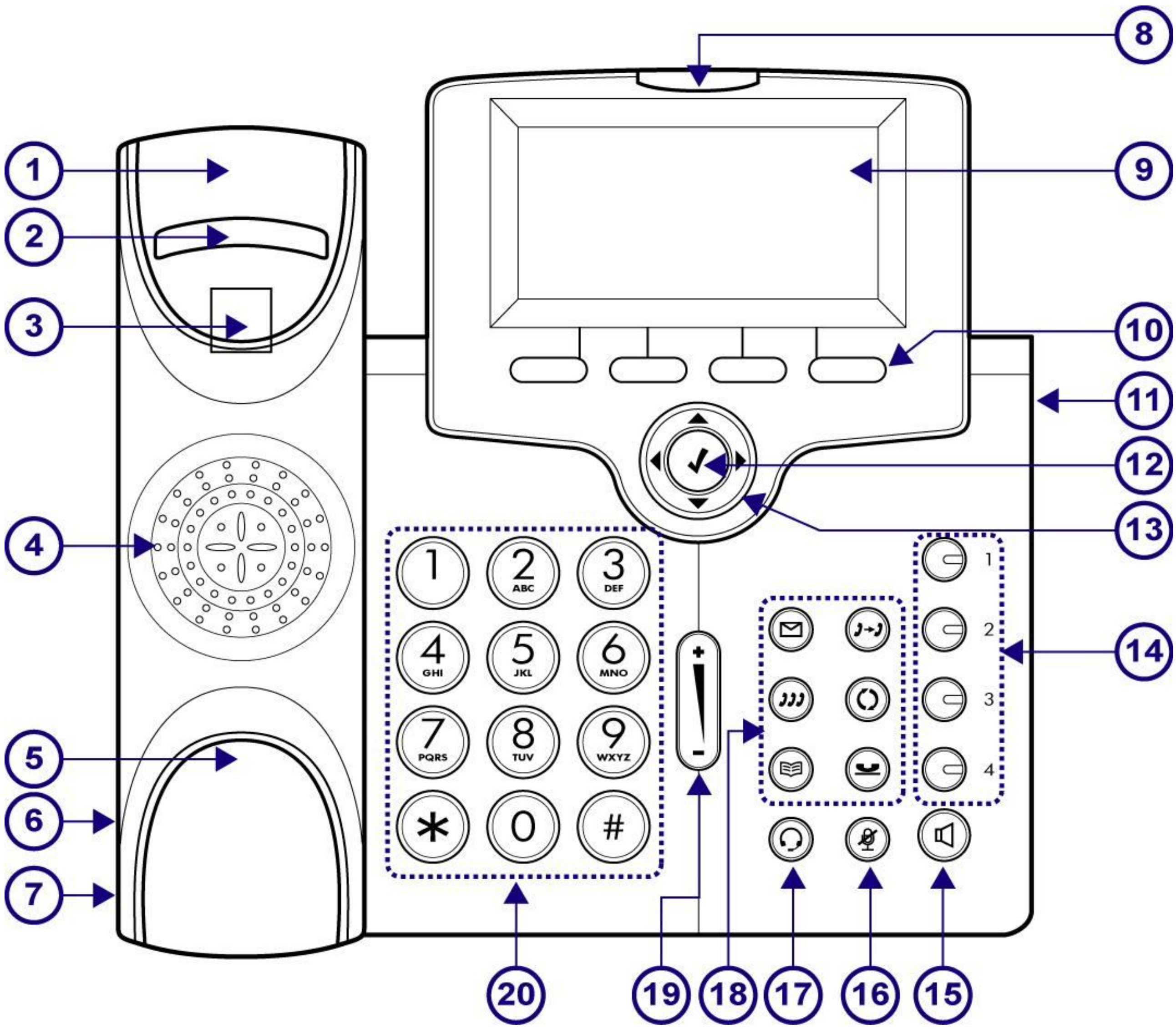
The IP2061 can transfer and receive voice via IP network. Therefore, it can be deployed and connected all over the world among headquarter and remote branch offices. Since it is a stand-alone and “always-on” terminal, there is no need to have any active PC to let it work. The IP2061 is completely stand-alone.

The IP2061 comes with a graphic LCD display, traditional keypad, several function keys, handset, I/O ports, and PoE (Power-over- Ethernet) /Power adaptor. It can be installed and placed on the desktop or mounted on the wall.



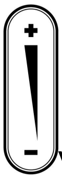
The Front-View of the IP2061

The figure below illustrates the front view of the IP2061. With the point numbers, you can find its name and a simple description of the part in the following table.



No	Part Name	Description
1	Handset top cradle	For the placement of handset receiver.
2	Hook switch	For hanging up your handset.
3	Cradle latch	For latching your handset when it is wall mounted.
4	Speaker	Provides sound for telephone ring and hands free talking.

5	Handset bottom cradle	For the placement of handset transmitter.
6	Handset cord port	Handset jack for connecting your handset.
7	Headset wire port	Headset jack for connecting your headset.
8	Message LED	Indicates an incoming call or new voice message.
9	Graphic LCD	Displays features such as the time, date, your phone number, caller ID, line/call status and soft key tabs.
10	Soft keys	Each activates a softkey option (displayed on your phone screen). Soft keys point to feature options displayed along the bottom of your LCD screen. Soft keys change depending on the status of your phone.
11	EDM	Connects to an Extension Dial Module.
12	Navigator [✓] OK Key	Used to confirm the setting or phone number dial.
13	Navigator Control Keys	The four arrows ◀ ▶ ▲ ▼ enable you to scroll through text and select features displayed on the LCD screen.
14	1,2,3,4 Line Keys	These keys can be used for line selection or programmable features. A green LED is associated with each key to indicate its line/call status.
15	SPKR key	Toggles the speaker on or off. A green LED is associated to indicate its status.
16	MUTE key	Toggles the mute on or off. A red LED is associated to indicate its status.
17	Headset key	Toggles the headset on or off. A green LED is associated to indicate its status.
18	XFR key	Transfer a call to another IP phone.
	REDIAL Key	To redial the dialed number from Dialed Record menu list automatically.
	HOLD Key	Place the current call on hold so you may place another phone call.
	MSG Key	To get access to Voice Mail System for message retrieval.

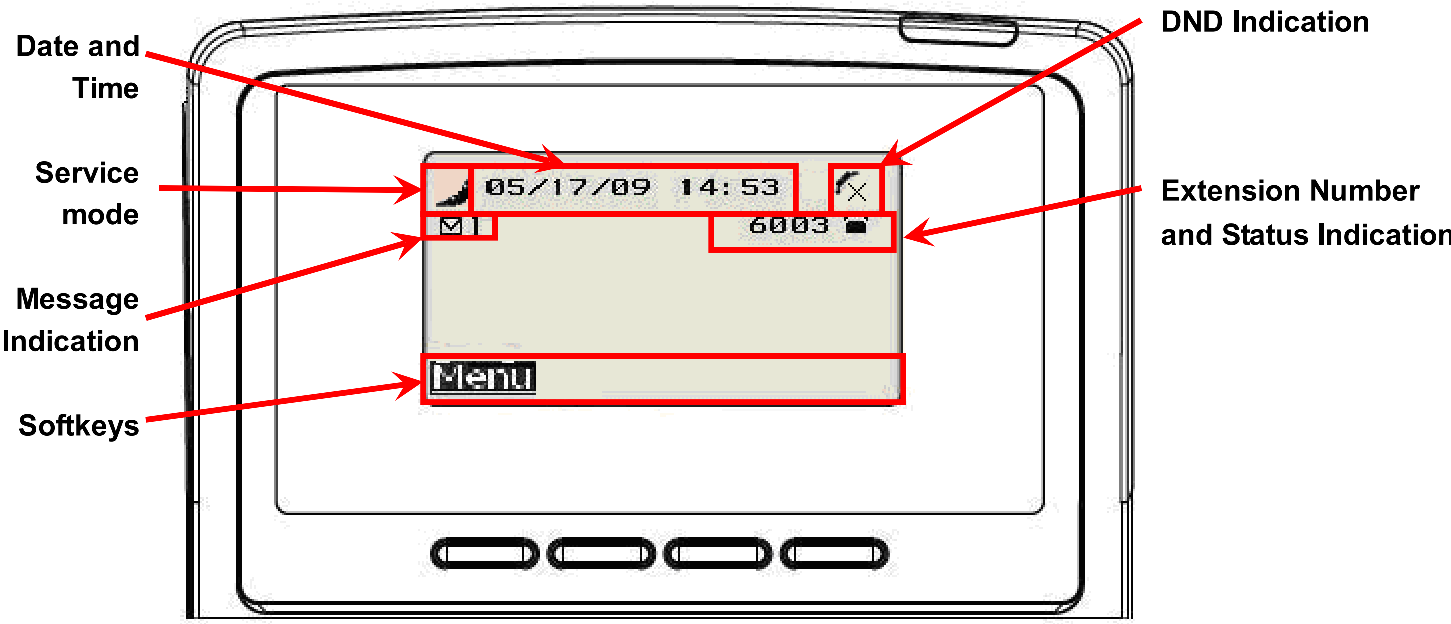
	CONF Key	To initiate a conference call after multiple calls are connected.
	Phonebook Key	Access the Phone Book so you may call or edit the contacts.
19	Volume Control Key	Increases or decreases volume for the handset, headset, or speakerphone (depending upon which is currently active). Also controls the ringer volume (if the handset is in its cradle)
20	Numeric Keypad	[1], [2]… [9], [*], [0], [#]: The numeric keypad for dialing numbers.

The LED Indication

LED	Color	Status	Description
MSG	Red	Off	Idle or no new message.
		Blinking Slowly	New voice message indication.
		Blinking Rapidly	There is an incoming call.
		Blinking Continuously	IP2061 cannot register to IG6600.
SPKR key	Green	Off	The hands-free speaker is not in use.
		Steady	While in on-hook dialing mode or hands-free talking mode.
MUTE key	Red	Off	The microphone is active for handset, headset or hands-free mode.
		Steady	The microphone is inactive for handset, headset or hands-free mode.
Headset key	Green	Off	The headset mode is disabled.
		Steady	The headset mode is enabled.
1,2,3,4 Trunk lines keys	Green	Off	The trunk line is un-activated or idle.
		Steady	The trunk line is active (dialing, or during a call).
		Blinking Slowly	The call of relative trunk line is on hold.
		Blinking Rapidly	There is an incoming call from that trunk line.

The LCD Indication

The following figure shows a standard format of LCD screen. There are 4 soft keys associated with the operation of LCD display. For different menu or status, the display format will be changed accordingly.
The LCD also supports to have the back light.



Date and Time: If the phone is registered to the IG6600 via Plug-n-Play, the IG6600 will send the info to the phone; if the phone is registered to the IG6600 remotely and Network Time Server is set, phone will sync the correct time to time server according to user's time zone setting; else, it shows the passed time since latest boot.

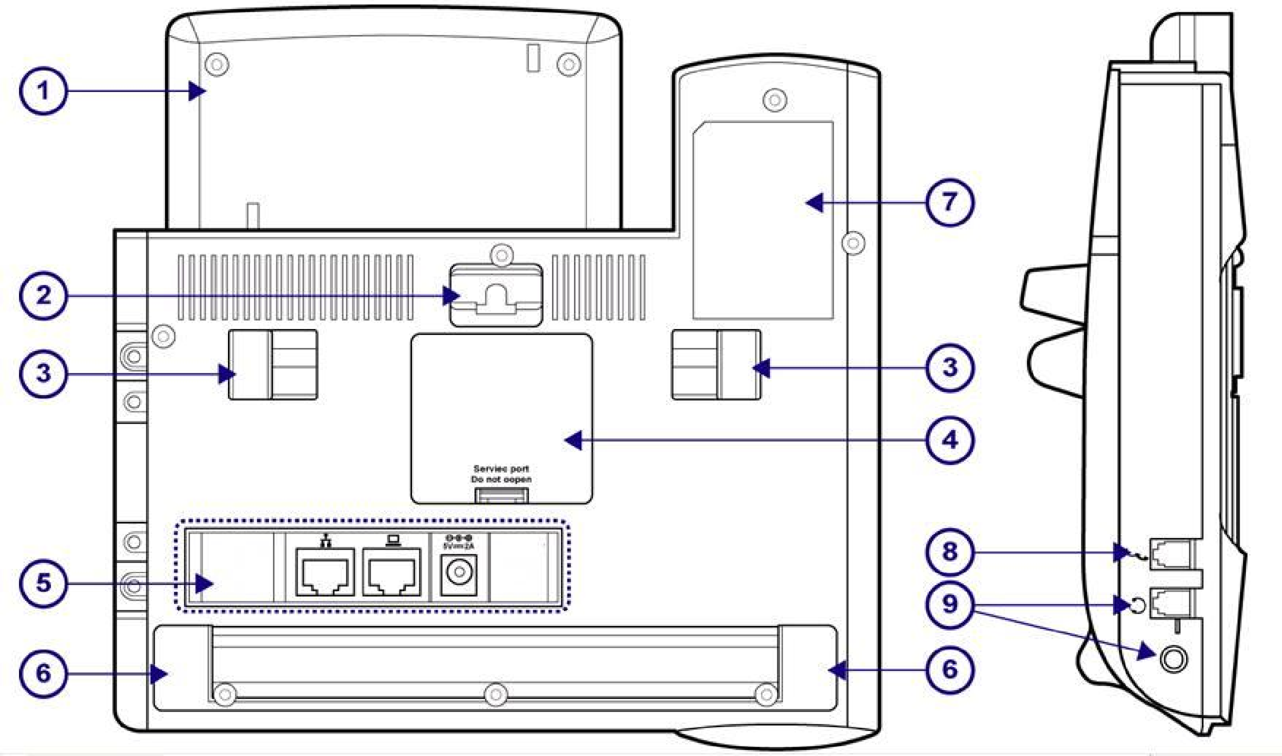
Message Indication: If there are voice messages left on the IG6600, the phone will show the number of unread messages.

DND Indication: If DND (Do Not Disturb) function is enabled, the phone will show DND icon on LCD.

Service Mode: When service mode is night mode, the phone will show moon icon on LCD.

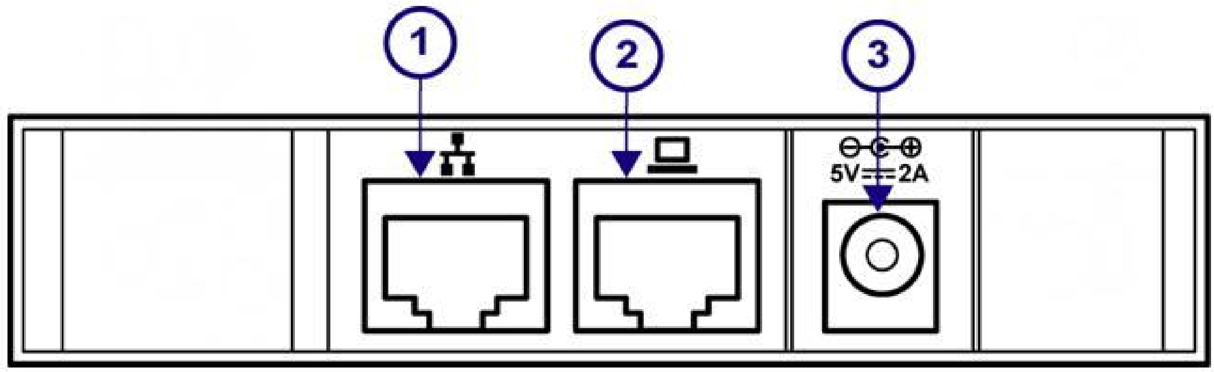
Extension Number and Status Indication: There are three line statuses, registered, un-registered and phone's always call forward. For registered status, phone will show icon after line number. For un-registered status, phone will show icon after extension number. When the phone is registered, if Call Forward Direct function is enabled, icon will replace the original registered icon.

The Rear-View of the IP2061



No	Part Name	Description
1	**LCD Screen Cover**	Covers the LCD screen.
2	**Wall-Mount Hole**	For hanging the phone on the wall.
3	**Hinge of Stand**	For supporting the IP phone at different angles. For wall mounting, please remove the stands.
4	**Service Door**	For service only.
5	**Input /Output Ports**	IP phone connectors (see the next figure for the details).
6	**Hinge of Chassis**	For supporting the IP phone at different angles.
7	**Product Label**	To show product production information, such like Product model, serial number and MAC address.
8	**Handset Cord Port**	Handset cord jack on the side of IP phone.
9	**Headset Wire Port**	(1) Headset wire jack on the side of IP phone. (2) Smaller Headset jack for different plug type.

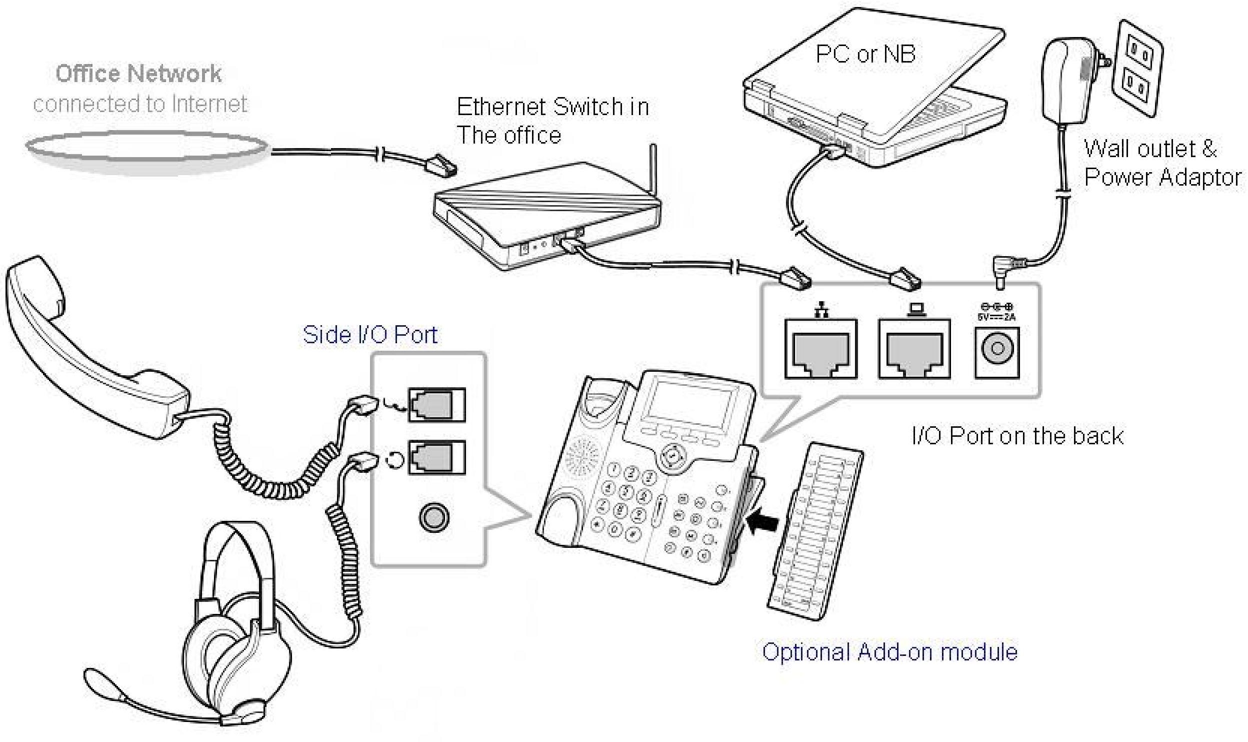
Understanding the Connectors of the IP Phone



No	Part Name	Description of function
1	**LAN Port**	Ethernet port for connecting to IP network.
2	**PC Port**	Ethernet port for connecting to PC or Notebook PC. This port can also be link to the other IP-Phones as P2P intercom function.
3	**Power Jack**	If a power source from adaptor is required, please use a standard power adaptor supplied in the package.

Hardware Installation

Before using phone, you have to prepare your network environment first. Please see the figure below. You can find an Ethernet switch (or Router or Hub) and connect it to your office network that is accessible to the global Internet. For sure, your network environment has also a regular DHCP server that can offer you one applicable IP address to the IP phone.



Then, you can start to install the phone.

1. Plug the optional add-on Module (IP-EDM) onto the IP phone) and fasten it with 2 screws.
2. Connect one end of the coiled telephone cord to the handset and the other end tp the handset port on the side of IP phone.
3. Connect an Ethernet cable to the LAN Port of your switch/router/hubs.
4. Then connect the other end of this Ethernet cable to the LAN port of the IP phone.
5. For local configuration purpose on the web of IP phone, you may connect a PC or notebook via another Ethernet cable to the PC port on the back of IP phone.
6. Plug the power cord to the power jack of IP Phone and then plug the power adapter to the wall outlet. Remember to power on your Ethernet switch/router/hub as well.
7. Wait for your phone to boot up. Please do not interrupt this process as it may take a few minutes.
8. When the prompt on the LCD display is ready, please check if the IP phone works by lifting up the handset. If there is a dialing tone, then dial several numbers to see whether the LCD screen is showing your dialed numbers or not.
9. Check all the connections and reconfiguration of the IP phone, if you do not hear a dial tone.

3. General Operations

Introduction

To operate the IP2061 with IG6600, you need to know some conventions that will be mentioned in this manual. In the following descriptions, we will introduce some terminologies for your understanding.

Register to IG6600

When IP2061 firstly connecting to LAN side of IG6600 or at WAN side but in the same subnet, IG6600 will assign an unused phone number to IP2061. After that, even if IP2061 reboot, IG6600 will assign the same phone number to it. So IP2061 can register to IG6600 automatically in the following times.

Calls

The “Call” in this manual represents a connection with outside party. IP2061 supports 4 simultaneous calls, i.e. IP2061 can use 4 channels at the same time. IP2061 can dial the destination phone number directly for making a phone call. IP2061 supports 4 line keys. User can press line key to choose IP or PSTN trunks which the IG6600 registered to make an outgoing call. User can also dial IP or PSTN trunk access number for making an outgoing call. User can hold one call and talk to the other. Therefore, the IP2061 is said to support multiple-call appearance.

Caller ID & User ID

If the caller didn’t choose to hide his number and if the network supports the Caller ID feature, the caller's phone number is shown on the screen when you receive a call. If the caller choose to hide his number or the network doesn’t support the Caller ID feature, the IP2061 will display the user’s ID of the caller if it is available.

To Install the IP2061

Before the operation of IP2061, you have to install the phone well into the network. Please refer to previous section “*The Rear-View of the phone*”. Connecting the LAN port to IG6600 or to hub/switch with an Ethernet cable, then connect the handset to handset port with a cord. After that, you could plug the power adaptor to power port, the phone will switch on and work normally.

To Configure Your IP2061 for Service

Furthermore, you have to configure the phone well before operation. You may refer to this administrator manual for full information on how to configure all the settings of the IP2061.
Now, if the IP2061 is already connected to IG6600, please follow the following chapters to

operate the phone.

The configuration menu to configure IP2061 is as follows:

You may navigate through the menu with the navigation keys. The following sections will describe how you can setup your IP2061 through this menu.

Here lists several meanings of icons for you to easily understand the call features:

Off-hook On-hook Talk Keypad

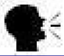
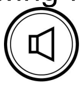
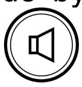
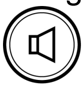
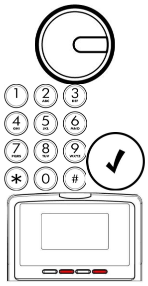
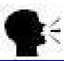
Navigation keys ENTER key Volume control key

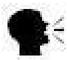
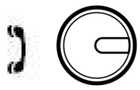
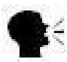
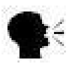
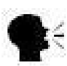
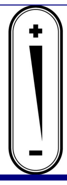
REDIAL key MESSAGE key PHONEBOOK key TRANSFER key

CONFERENCE key HOLD key

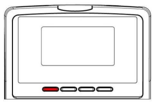
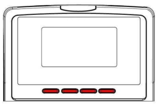
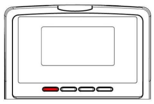
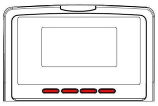
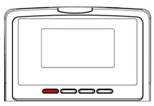
HEADSET key MUTE key SPEAKER key Line key

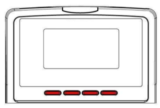
Basic Call Features

Operation	***Description***	
Making a Call		**1.** Pick-up the handset. You will hear dial tone played.
		**2.** Use the keypad to dial the phone number. Press ENTER key or “Dial” softkey to dial out immediately, or wait for a while (Dial Timeout setting) for auto-ending the dial. Phone will send out the dialed number to IG6600. IG6600 will route the call to destination. Note: You could use the “Backsp” softkey to delete the last digit.
		**3.** Start talking to called party.
		**4.** On-hook the handset when your conversation is over.
	There are three mode to establish a call for this IP phone, handset mode, hands-free mode and headset mode. In this manual, only handset mode is used for example. In the following features, you can replace  to hands-free mode by pressing  key or headset mode by pressing  and  key. When the conversation is over, press  key again to release call. For headset mode, please prepare a headset first.	
Making a Call via Specific Trunk		**1.** Pick-up the handset. You will hear dial tone played.
		**2.** Press the line key. The LED of line key will light up and the specific trunk line will be engaged. Use the keypad to dial the target phone number. Press # key to dial out immediately, or wait for a while (Dial Timeout setting) for auto-ending the dial. **3.** User can also dial the PSTN or IP trunk access number which the IG6600 registered. For detail about IP and PSTN trunk, refer to IG6600’s manual. Note: You could use the “Backsp” softkey to delete the last digit.
		**4.** Start talking to called party.

		5. On-hook the handset when your conversation is over.
Receiving a Call		1. Pick-up the handset while hearing phone's ringing.
		2. Start talking to caller party.
		3. On-hook the handset when your conversation is over.
Receiving a Call via Specific Trunk		1. While getting an incoming call from a specific trunk, the relative line key LED will flash and the phone rings. You can pick-up the handset or press the relative line key to receive the call.
		2. Start talking to caller party.
		3. On-hook the handset when your conversation is over.
Last Number Redial		4. Press the REDIAL key, the LCD will show last 30 dialed numbers.
		5. Use the navigation keys to select a dialed call and press "Dial" softkey to redial.
Mute the Microphone		1. While being engaged in a conversation (handset, headset or hands-free mode), you could mute the microphone by pressing the MUTE key.
		2. The LED of the MUTE button will light up. At this moment, the user may speak freely, the outside party will not hear anything.
Adjust the Voice Volume During a Conversation		1. During a conversation, if the voice volume is too low or too high, you may adjust it.
		2. Press the volume control key to adjust the volume.

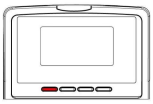
Call Log

Operation	***Description***	
Review Dialed Calls		1. Press the "Menu" softkey. (When IP2061 is idle, you can also press ▲ to enter the *Call Log* item directly)
	▲ ▼ **➔ Call Log**	2. Use the navigation keys to select the *Call Log* item. Press ENTER key to validate the selection.
	▲ ▼ **➔Dialed Calls**	3. Select the *Dialed Calls* item and validate with the ENTER key.
	▲ ▼ 	4. Use the navigation keys to review the dialed calls. You may choose to redial the number (using the "Dial" softkey). Press the "Del" softkey to delete the selected record. Press the "DelAll" softkey to delete all records. Press the "Cancel" softkey to exit the menu.
Review Received Calls		1. Press the "Menu" softkey. (When IP2061 is idle, you can also press ▲ to enter the *Call Log* item directly)
	▲ ▼  **➔ Call Log**	2. Use the navigation keys to select the *Call Log* item. Press ENTER key to validate the selection.
	▲ ▼ **➔Received Calls**	3. Select the *Received Calls* item and validate with the ENTER key.
	▲ ▼ 	4. Use the navigation keys to review the received calls. You may choose to redial the number (using the "Dial" softkey). Press the "Del" softkey to delete the selected record. Press the "DelAll" softkey to delete all records. Press the "Cancel" softkey to exit the menu.
Review Missed Calls		1. Press the "Menu" softkey. (When IP2061 is idle, you can also press ▲ to enter the *Call Log* item directly)
	▲ ▼ **➔ Call Log**	2. Use the navigation keys to select the *Call Log* item. Press ENTER key to validate the selection.

	▲ ▼ **➔Missed Calls**	3. Select the *Missed Calls* item and validate with the ENTER key.
	▲ ▼ 	4. Use the navigation keys to review the missed calls. You may choose to redial the number (using the “Dial” softkey). Press the “Del” softkey to delete the selected record. Press the “DelAll” softkey to delete all records. Press the “Cancel” softkey to exit the menu.

Information about the IP2061

You may view all related information about the IP2061 through the LCD menu. This may give you, for example, the current network settings of the IP2061.

Operation	*Description*	
View Information about the IP2061		1. Press the “Menu” softkey.
	▲ ▼ ➔ **Info**	2. Use the navigation keys to select the *Info* item and validate with the ENTER key.
	▲ ▼	3. Use the navigation keys to choose the information you would like to review.

The following information can be reviewed from the LCD screen of your IP2061:

- Model Name
- Firmware Number
- MAC Address
- Network Type
- IP Address
- Subnet Mask
- Default Gateway
- DNS Server

4. Advanced Operations

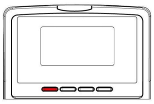
Network Settings

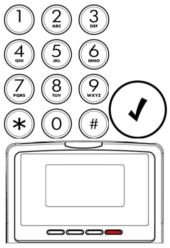
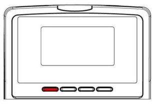
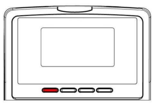
The default network settings are the following:

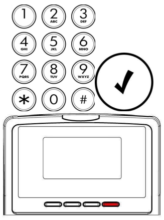
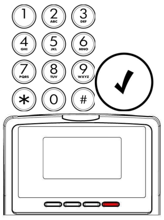
Default IP address/ Subnet Mask: Depend on DHCP server
Default Gateway: Depend on DHCP server
Default DNS: Depend on DHCP server

Default Administrator's Username of Web: admin
Default Administrator's Password of Web: 1234

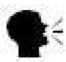
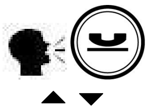
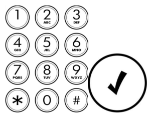
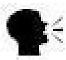
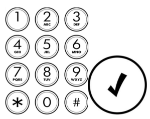
If you need to change these default settings, please refer to the following instructions.

Operation	***Description***	
Static IP Address		1. Press the "Menu" softkey.
	▲ ▼ **➔ Admin**	2. Use the navigation keys to select the *Admin* item. Press ENTER key to validate the selection. Type the administrator's password to get into *Admin* menu.
	▲ ▼ **➔ Network**	3. Use the navigation keys to select the *Network* item. Press ENTER key to validate the selection. The LCD displays the Network Settings menu.
	▲ ▼ **➔ Network Type**	4. Select the *Network Type* item and press ENTER key.
	▲ ▼ **➔ Static IP**	5. Use the navigation keys to choose *Static IP* type and press ENTER key to validate.
	▲ ▼ **➔ Static IP**	6. Go to previous menu, choose *Static IP* item and press ENTER key.
	▲ ▼ **➔ IP Address, Subnet Mask, Default Gateway, DNS**	7. There are *IP Address*, *Subnet Mask*, *Default Gateway*, and *DNS* items for user to configure.

		**8.** Use the keypad to enter the IP address, subnet mask, default gateway, or DNS of your IP2061. Use the * key for representing a dot •. Press ENTER key to validate the entered value. Note: You could use the "Backsp" softkey to delete the last character.
	You must reboot the phone to validate the network parameter changing.	
Dynamic IP Address (DHCP)		**1.** Press the "Menu" softkey.
	▲ ▼ **➔ Admin**	**2.** Use the navigation keys to select the *Admin* item. Press ENTER key to validate the selection. Type the administrator's password to get into *Admin* menu.
	▲ ▼ **➔ Network**	**3.** Use the navigation keys to select the *Network* item. Press ENTER key to validate the selection. The LCD displays the Network Settings menu.
	▲ ▼ **➔ Network Type**	**4.** Select the *Network Type* item and press ENTER key.
	▲ ▼ **➔ DHCP**	**5.** Use the navigation keys to choose *DHCP* type and press ENTER key to validate.
	You must reboot the phone to validate the network parameter changing.	
PPPoE		**1.** Press the "Menu" softkey.
	▲ ▼ **➔ Admin**	**2.** Use the navigation keys to select the *Admin* item. Press ENTER key to validate the selection. Type the administrator's password to get into *Admin* menu.
	▲ ▼ **➔ Network**	**3.** Use the navigation keys to select the *Network* item. Press ENTER key to validate the selection. The LCD displays the Network Settings menu.
	▲ ▼ **➔ Network Type**	**4.** Select the *Network Type* item and press ENTER key.

▲ ▼ ➔ **PPPoE**	**5.** Use the navigation keys to choose *PPPoE* type and press ENTER key to validate.
▲ ▼ ➔ **PPPoE**	**6.** Go to previous menu, choose *PPPoE* item and press ENTER key.
▲ ▼ ➔ **Username**	**7.** Choose *Username* item and press ENTER key.
	**8.** Use the keypad to enter correct PPPoE username for PPPoE mode. Press ENTER key to validate the input. Note: You could use the “Backsp” softkey to delete the last character.
▲ ▼ ➔ **Password**	**9.** Go to previous menu, choose *Password* item and press ENTER key.
	**10.** Use the keypad to enter correct PPPoE password for PPPoE mode. Press ENTER key to validate the input. Note: You could use the “Backsp” softkey to delete the last character.
You must reboot the phone to validate the network parameter changing.	

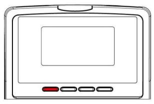
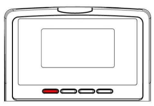
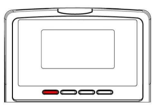
Advanced Call Operations

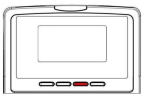
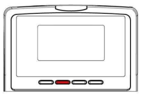
Operation		Description
Call Hold		1. While being engaged in a conversation, you may hold a call.
		2. Press HOLD key to hold the call.
		3. To resume a held call, simply press the HOLD key again.
		4. If a trunk line call is held, you can also press the line key to resume the trunk line call.
3-Way Conference Call		1. While being engaged in a conversation, you may invite another party to join using the 3-way conference call feature. To achieve this, press the HOLD key to hold the first call. Use the navigation keys to choose an unused channel.
		2. Use the keypad to dial the phone number. Press ENTER key to validate, the IP phone will call out the other party.
		3. You can talk with the second party prior to let him/her join the conference.
		4. When you are ready to make the conference, press the CONFERENCE key. The LCD screen will show "CONF" that the two calls are in a conference.
Call Transfer (Blind Transfer)		1. While being engaged in a conversation, press the TRANSFER key. The phone prompts you to enter the phone number which you would like to transfer the call to.
		2. Use the keypad to dial the phone number. Press ENTER key to validate, the IP phone will transfer the call to other party.
		3. You can hang up the handset now. The call has already been transferred.

Call Transfer (Attended Transfer)		1. While being engaged in a conversation, hold the call by pressing the HOLD key.
		2. Use the navigation keys to choose an unused channel. Use the keypad to dial the phone number. Press ENTER key to validate, the IP phone will call out the other party.
		3. You can talk with the transfer target prior to transfer the call.
		4. When you are ready to transfer the call, press the TRANSFER key.
		5. You can hang up the phone now. The call has already been transferred.

Phonebook

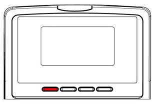
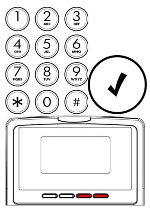
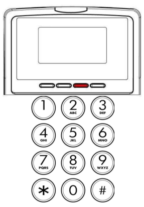
The phonebook feature let's you store a list of phone numbers. (For IP2061, it could be stored up to 200 entries)

Operation	***Description***	
Phonebook (Browse and Dial a Number)	 ▲ ▼ **➔ Phone Book**	**1.** Press the "Menu" softkey. **2.** Use the navigation keys to select the *Phone Book* item and press ENTER key.
		**3.** Or just press PHONEBOOK key to enter phonebook menu.
	There are 4 items shown, *Item*, *Name*, *Group*, and *New*.	
	▲ ▼ **➔ Item**	**4.** To browse the phonebook data by "Index", select *Item* and press ENTER key. Then use navigation keys to browse all phonebook data.
	▲ ▼ **➔ Name**	**5.** To browse the phonebook data by "Name", select *Name* and press ENTER key. Then use navigation keys to browse.
	▲ ▼ **➔ Group**	**6.** To browse the phonebook data by "Group", select *Group* and press ENTER key. There are 5 groups for user to classify all phonebook data. Select an item and press ENTER key. Then use navigation keys to browse.
		**7.** When browsing certain phonebook data, press "Dial" softkey to dial out the phonebook number.
Phonebook (Add, Edit, or Delete a number)	 ▲ ▼ **➔ Phone Book**	**1.** Press the "Menu" softkey. **2.** Use the navigation keys to select the *Phone Book* item and press ENTER key.
		**3.** Or just press PHONEBOOK key to enter phonebook menu.
	▲ ▼ **➔ New**	**4.** Select *New* and press ENTER key. You can enter Number first. After enter Number and press Enter key, you can press "Edit" softkey to edit other items. There are 4 items for user to

		add, Name, Number, Ring Type and Group.
		5. When browsing certain phonebook data, press "Edit" softkey to edit. There are 4 items for user to edit, Name, Number, Ring Type and Group.
		6. When browsing certain phonebook data, press "Del" softkey to delete that data.

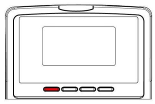
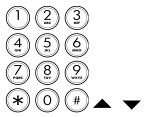
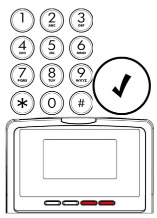
Speed Dialing

The Speed Dialing feature let's you store a list of 10 phone numbers that you can access easily using a Speed Dialing number from 0 to 9.

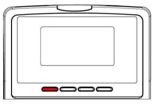
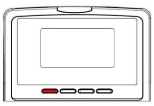
Operation	***Description***	
Speed Dialing (Add, Edit, or Delete a Number)		**1.** Press the "Menu" softkey. (When IP2061 is idle, you can also press ▼ to enter the *Speed dialing* directly)
	▲ ▼ ➔ **Speed Dialing**	**2.** Use the navigation keys to select the *Speed Dialing* item and press ENTER key.
	▲ ▼	**3.** You can use the navigation keys to browse all the Speed Dialing numbers.
		● If you want to add a new number, find an empty position, then press "Edit" softkey and using keypad to enter a number. Press ENTER key to validate. ● If you want to edit an existing number, find the number you would like to edit. Press "Edit" softkey. Use the "Backsp" softkey to clear the number and use keypad to enter a new number. Press ENTER key to validate.
Dial a Speed Dialing Number		**1.** Pick up the handset.
		**2.** Press the "SPD" (Speed Dialing) softkey. Then enter a SPD number between 0 and 9. The corresponding phone number will be dialed out.

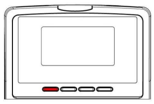
Blocking List

The IP2061 provides the possibility to block calls for a specific Caller ID. You can input at most 10 blocking entries.

Operation	Description	
Caller Blocking (Add, Edit, or Delete a Number)		**1.** Press the "Menu" softkey.
	▲ ▼ ➔ **Call Blocking**	**2.** Use the navigation keys to select the *Call Blocking* item to enter the Call Blocking settings menu. You can use the navigation keys to select *Status* item and turn on or off the call blocking function.
	▲ ▼ ➔ **Blocking List**	**3.** Use the navigation keys to select the *Blocking List* item to enter the Caller Blocking settings menu. The LCD screen prompts you to enter a number between 0 and 9.
		**4.** You may enter the desired number directly using the keypad or use the navigation keys to browse all the blocked Caller ID.
		● If you want to add a new number, find an empty position, then press "Edit" softkey and using keypad to enter a number. Press ENTER key to validate. ● If you want to edit an existing number, find the number you would like to edit. Press "Edit" softkey. Use the "Backsp" softkey to clear the number and use keypad to enter a new number. Press ENTER key to validate.

Call and Phone Management

Operation	Description	
Call Forward		**1.** Press the “Menu” softkey.
	▲ ▼ ➔ **Call Forward**	**2.** Use the navigation keys to select the *Call Forward* item and press the ENTER key to validate the selection. Select the next item according to the type of call forward (*Always Forward*, *Busy Forward*, *No Answer Forward* or *DND forward*) you would like to activate.
	▲ ▼ ➔ **Status**	**3.** Choose the *Status* item to turn On/Off the call forwarding feature. Select On or Off then press the ENTER key to validate.
	▲ ▼ ➔ **Forward Number**	**4.** Select the *Forward Number* item to input the phone number you wish to forward the calls to.
	▲ ▼ ➔ **Forward Call Type**	**5.** Select the *Forward Call Type* item to select ICM, Outside or Both. ICM means enable the call forward when the call comes from internal extension. Outside means enable the call forward when the call comes from IP trunk, PSTN trunk or IGW Group. Both means enable the call forward when the call comes from internal extension, IP Trunk, PSTN trunk or IGW Group.
	▲ ▼ ➔ **No Answer Forward**	**6.** For *No Answer Forward* item, there is one additional item “*No Answer Time*” for user to set. If the phone doesn’t answer the call after this time, the call will be forwarded to the destination. (Default value is 15 seconds)
	Note: It is possible to activate different kinds of call forward at the same time. Ex: *No Answer forward* + *Busy Forward*, etc.	
Auto Answer	You can use this function to auto answer all incoming call when you are busy on something and hope the call can be received automatically.	
		**1.** Press the “Menu” softkey.
	▲ ▼ ➔ **Auto Answer**	**2.** Use the navigation keys to select the *Auto Answer* item and press ENTER key.
	▲ ▼	**3.** Use the navigation keys to choose to turn On or Off and press ENTER key to validate.

DND	DND means "Do Not Disturb". You can enable this function if you don't want any incoming call to disturb your work. And all incoming call will get busy tone when they make call to your phone number.	
		1. Press the "Menu" softkey.
	▲ ▼ ➔ **DND**	2. Use the navigation keys to select the DND item and press ENTER key.
	▲ ▼	3. Use the navigation keys to choose to turn On or Off the DND feature and press ENTER key to validate. While enabling DND function, it shows DND icon on the LCD at idle state.
	Note: If both *DND* and *Call Forward* are turned on, the priority will be *DND Forward* > *Always Forward* > *Busy Forward* or *No Answer Forward*.	

5. Web Configuration

Login Information

The default network settings are the following: **Default IP address / Subnet Mask: Depend on DHCP server** **Default Gateway: Depend on DHCP server** **Default DNS: Depend on DHCP server** **Default Administrator's Username of Web: admin** **Default Administrator's Password of Web: 1234** **Default User's Username of Web: user** **Default User's Password of Web: 1111**

There are two modes to access the IP phone's webpage: Administrator and User mode. Here lists several differences between two modes:

1. There is no Network page for user mode.
2. There is no SW upgrade page for user mode.
3. In Phone page of user mode, the handset/handsfree/headset Mic and all voice parameter items are hidden.
4. For user mode, most of SIP settings are hidden, only Call Forwarding/Caller Blocking Settings are kept for user to set.
5. In System page of user mode, only user's password, time settings and reboot phone are kept for user to set.

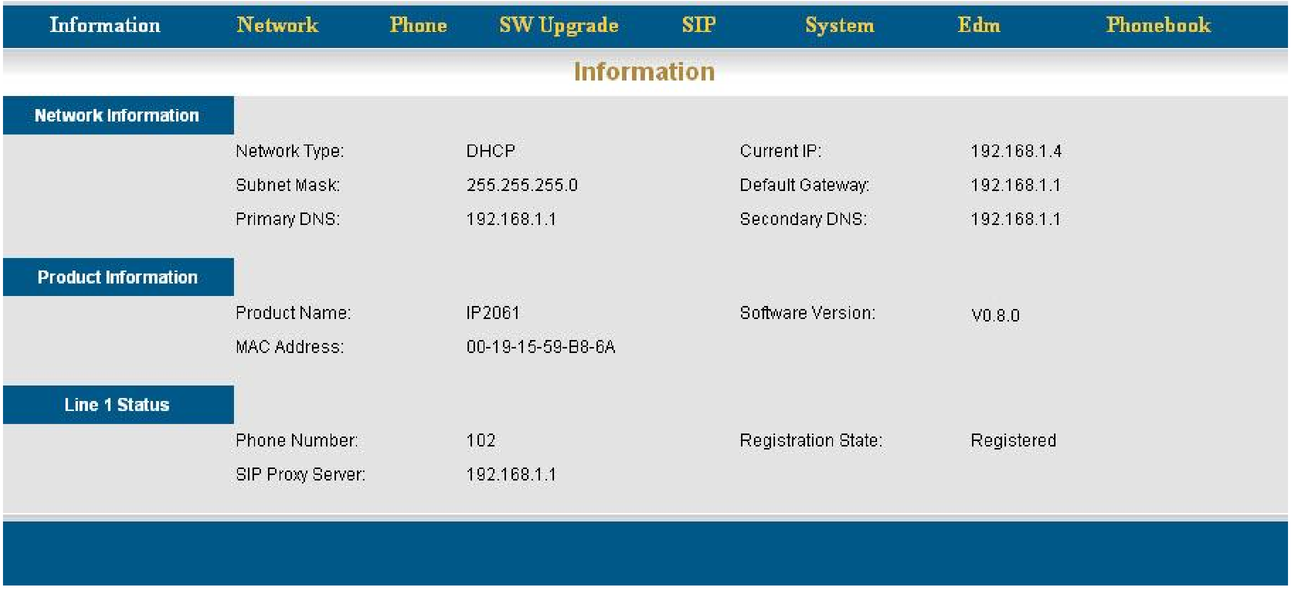
Accessing the phone through web browser, just simply enter the "**http://phone-ip**" in the location field of the browser. If you are not sure about the IP address of the phone, you can examine the current IP address through phone's menu. (Press "Menu" softkey, select item "6. Info", use navigation keys to check the IP address)

Press ENTER key to validate, the above dialog box will pop up and prompt you to provide the username and password in order to prevent unauthorized user accessing the phone. Please input the username and password then press OK button to enter phone's webpage.

Configuration Pages

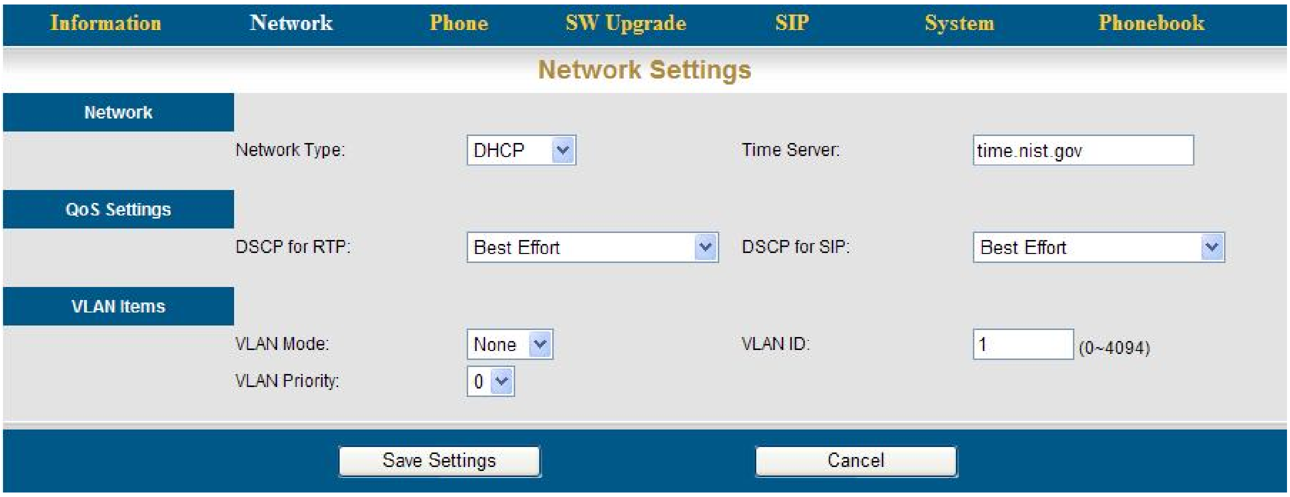
Information Page

The following is the default page you will see when you login to the phone’s webpage.



In this page, it shows Network/Product information and Line status.

Network Page



Network:

Network Type	Select the network type. There are three types: DHCP, Static IP, or PPPoE.
Time Server	SNTP (Simple Network Time Protocol) server. Phone will sync correct time from the server according to phone's time zone setting (In "System" page).
IP Address	Static IP address.
Subnet Mask	Static Subnet Mask.
Default Gateway	Static Default Gateway.
Primary DNS	Static Primary DNS.
Secondary DNS	Static Secondary DNS.
Third DNS	Static Third DNS.
PPPoE Username	Username for PPPoE mode.
PPPoE Password	Password for PPPoE mode.

The Primary/Secondary/Third DNS, Subnet Mask, and Default Gateway are used for Static IP mode, not for DHCP/PPPoE mode.
It is necessary to reboot the phone after you change the above network related items. Press "Save Settings" button first, then go to "System" page and press "Reboot" button to restart the phone.

QoS Settings:

DSCP for RTP	Select Differentiated Services Code Point (DSCP) for RTP.
DSCP for SIP	Select Differentiated Services Code Point (DSCP) for SIP.

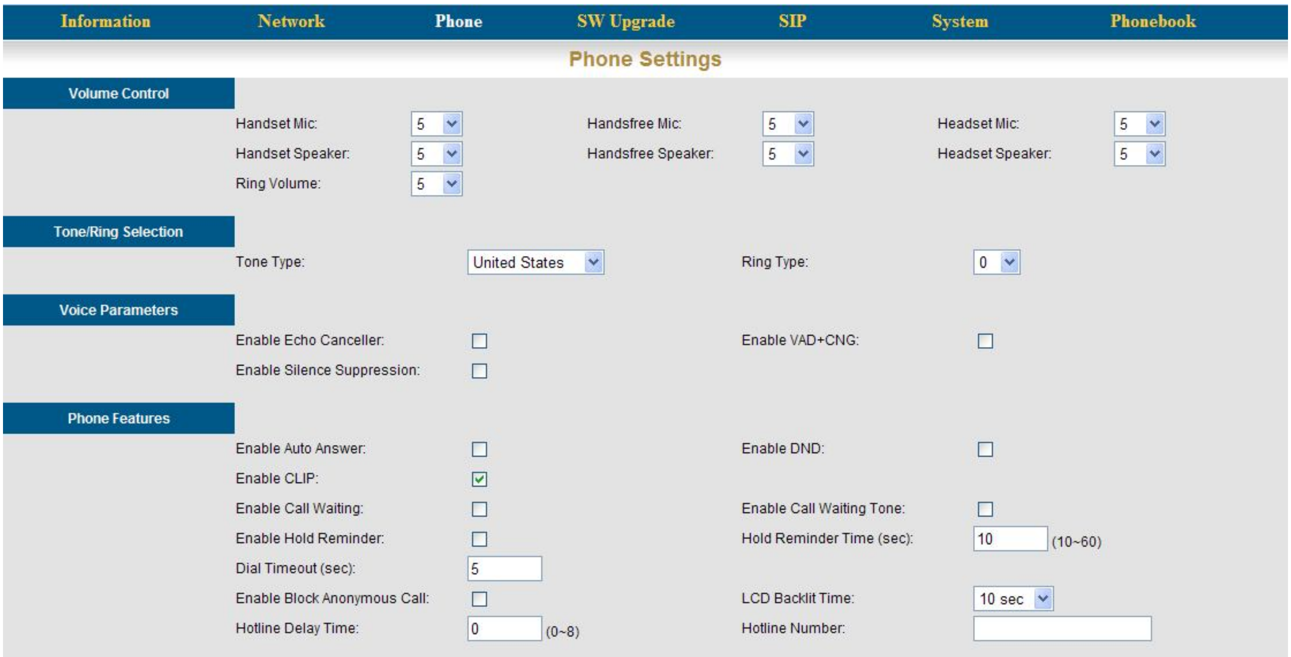
VLAN Items:

VLAN Mode	Select VLAN Routing Mode.
VLAN ID	Select VLAN Identification.
VLAN Priority	Select VLAN Priority Tag.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.

Phone Page



Volume Control:

Field Name	***Function***
Handset Mic	Select the volume level of handset microphone.
Handset Speaker	Select the volume level of handset speaker.
Handsfree Mic	Select the volume level of hands-free microphone.
Handsfree Speaker	Select the volume level of hands-free speaker.
Headset Mic	Select the volume level of headset microphone.
Headset Speaker	Select the volume level of headset speaker.
Ring Volume	Select the volume level of ring.

Tone/Ring Selection:

Tone Type	Select the tone type for country. There are 11 tone types for selection.
Ring Type	Select the ring type. There are10 ring types for selection.

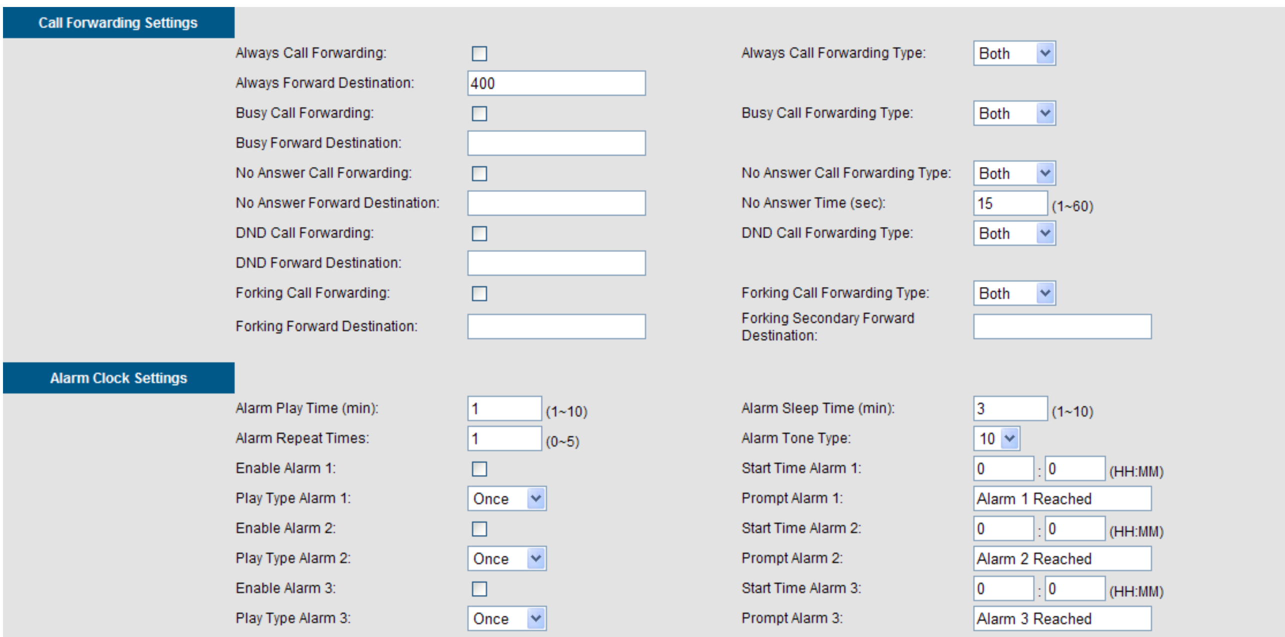
Voice Parameters:

Enable Echo Canceller	Turn on the Echo Cancellation function.
Enable VAD+CNG	Turn on VAD (Voice Activity Detection) and CNG (Comfortable Noise Generation) function.
Enable Silence Suppression	Turn on the Silence Suppression function.

Phone Features:

Enable Auto Answer	Turn on Auto Answer function. When it is enabled, phone will

	pick up the call automatically.
Enable DND	Turn on "Do Not Disturb" function. When it is enabled, all incoming call will get busy tone when they make call to this phone number. User can also press *4 (feature access code) to enable DND function.
Enable CLIP	Turn on the CLIP (Calling Line Identity Presence). The caller ID of incoming call will show on LCD.
Enable Call Waiting	When Call Waiting function is enabled and user is in talk state, phone can allow other incoming calls. It flashes the line key LED and Call Waiting Tone (If Call Waiting Tone is enabled) to indicate user. If it's disabled, when user is in talk state, other incoming caller will get busy tone. User can also press *98 (feature access code) to enable Call Waiting function.
Enable Call Waiting Tone	Turn on the Call Waiting Tone function. At talk state, a beep tone will indicate user every 10 seconds when another call is ringing in.
Enable Hold Reminder	Turn on the Hold Reminder function.
Hold Reminder Time (sec)	When a call is held, phone will remind user with a beep tone every this time period.
Dial Timeout (sec)	Set the timeout to automatically end dialing.
Enable Block Anonymous	Turn on to block the incoming call without Display Name.
LCD backlit Time	Set the LCD backlit time.
Hotline Delay Time	Set the delay time to activate the hotline application.
Hotline Number	Set the destination of the hotline application.



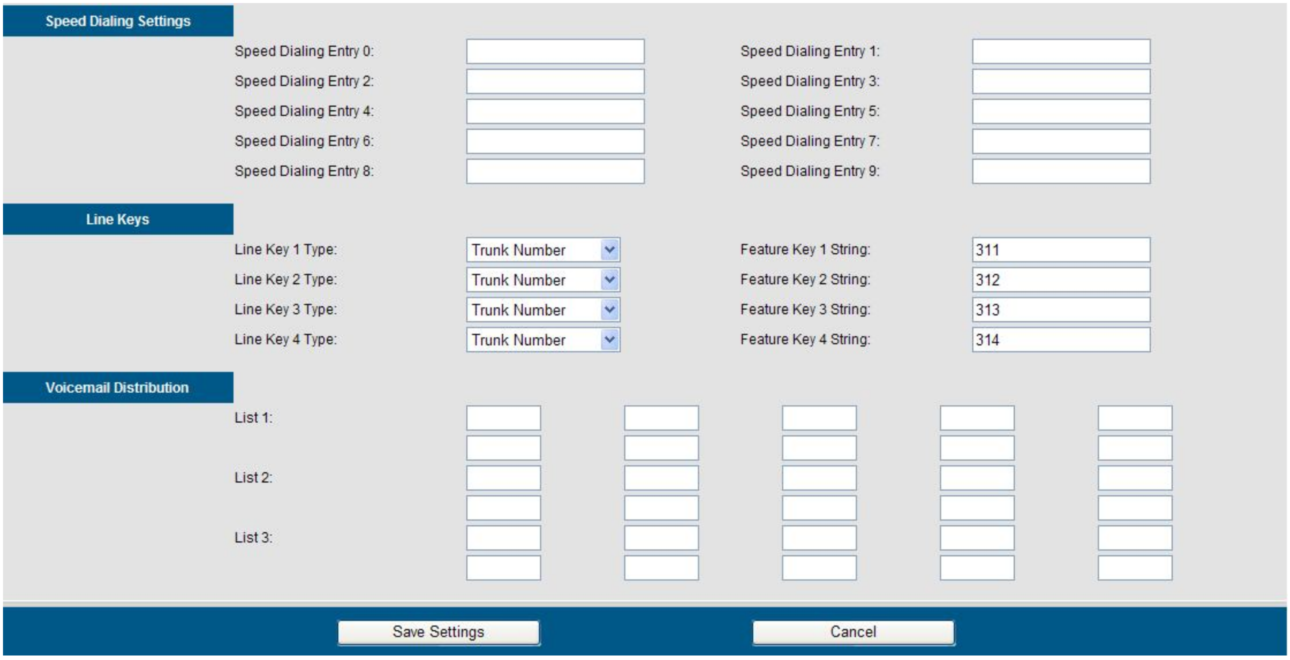
Call Forwarding Settings:

Always Call Forwarding; Always Forward Destination Always Call Forwarding Type	Turn on the always call forwarding function, the down side is where the call will be forwarded to. Always Call Forwarding Type can select the type of always call forwarding. Outside means when the call is from IP Trunk, PSTN Line or IGW Group, the function will active. ICM means when the call is from inner phone, the function will active. Both means the active will always active. To use the function by feature access code, refer chapter 6.
Busy Call Forwarding; Busy Forward Destination Busy Call Forwarding Type	Turn on the busy call forwarding function, the down side is where the call will be forwarded to. No Answer Call Forwarding Type can select the type of no answer call forwarding. Outside means when the call is from IP Trunk, PSTN Line or IGW Group, the function will active. ICM means when the call is from inner phone, the function will active. Both means the active will always active. To use the function by feature access code, refer chapter 6.
No Answer Call Forwarding; No Answer Forward Destination No Answer Call Forwarding Type	Turn on the no answer call forwarding function, the down side is where the call will be forwarded to. No Answer Call Forwarding Type can select the type of no answer call forwarding. Outside means when the call is from IP Trunk, PSTN Line or IGW Group, the function will active. ICM means when the call is from inner phone, the function will active. Both means the active will always active. To use the function by feature access code, refer chapter 6.
No Answer Timeout (sec)	Set the timeout for No Answer Call Forwarding.
DND Call Forwarding ; DND Forward Destination DND Call Forwarding Type	Turn on the DND forwarding function, the down side is where the call will be forwarded to. DND Call Forwarding Type can select the type of DND call forwarding. Outside means when the call is from IP Trunk, PSTN Line or IGW Group, the function will active. ICM means when the call is from inner phone, the function will active. Both means the active will always active. To use the function by feature access code, refer chapter 6.
Forking Call Forwarding: Forking Forward Destination: Forking Second Forward Destination Forking Call Forwarding Type:	Turn on the Forking forwarding function, the down side is where the call will be forwarded to. Forking Call Forwarding Type can select the type of Forking call forwarding. Outside means when the call is from IP Trunk, PSTN Line or IGW Group, the function will active. ICM means when the call is from inner phone, the function will active. Both means the active will always active. To use the function by feature access code, refer chapter 6.

Alarm Clock:

Alarm Play Time	Set the Play time when the alarm is reached.
Alarm Sleep Time	Set the Sleep time between Repeat Times
Alarm Repeat Times	Set the Repeat times. It’s applied if the alarm isn’t deactivated.

Alarm Tone Type	Set the specified ringing tone when the alarm is reached.
Enable Alarm 1	Enable the first alarm.
Start Time Alarm 1	Set the activated time of the first alarm.
Play Type Alarm 1	Set Play Time of the first alarm. It could be “once” or “Always”.
Prompt Alarm 1	Set the prompt of the first alarm. It will be shown on the LCD when the alarm is reached.
Enable Alarm 2	Enable the second alarm.
Start Time Alarm 2	Set the activated time of the second alarm.
Play Type Alarm 2	Set Play Time of the second alarm. It could be “once” or “Always”.
Prompt Alarm 2	Set the prompt of the second alarm. It will be shown on the LCD when the alarm is reached.
Enable Alarm 3	Enable the third alarm.
Start Time Alarm 3	Set the activated time of the third alarm.
Play Type Alarm 3	Set Play Time of the third alarm. It could be “once” or “Always”.
Prompt Alarm 3	Set the prompt of the third alarm. It will be shown on the LCD when the alarm is reached.



Speed Dialing Settings:

Speed Dialing Entry 0~9	There are 10 speed dialing entries for user to set.

Line Keys:

Line Key Type 1~4	There are 6 types could be chosen: None, Extension Number, Trunk Number, Park number, Feature Code and Others. Phone will do the relative function when user pressing related line key.

Feature Key String 1~4	The string will be executed when the line key type is set to Extension Number, Trunk Number, Park number, Feature Code and Others.

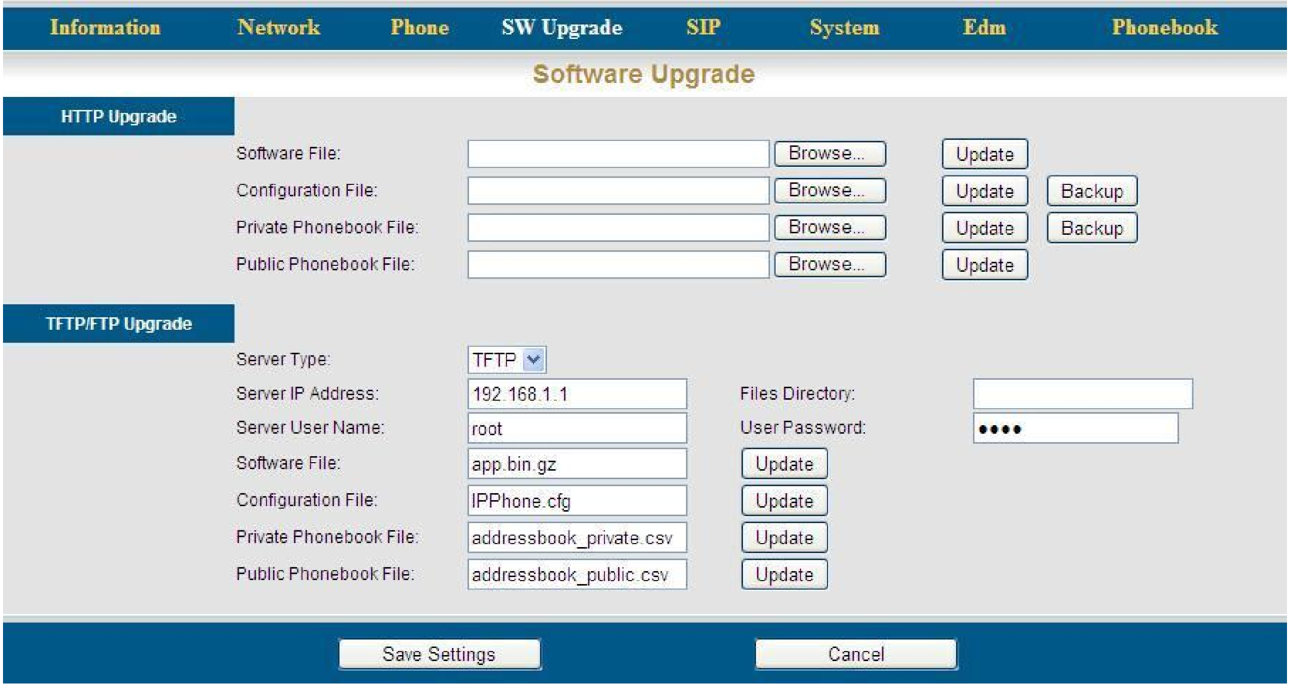
Voicemail Distribution:

List 1~3	There are 3 distribution lists for user to set. User can distribute the mailbox message to multiple mailboxes by assigning the distribution list.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.

SW Upgrade Page



HTTP Upgrade:

Field Name	***Function***
Software File	Select desired software and press Update button to upgrade software.
Configuration File	Select the desired configuration file and press Update button to restore the settings. Press Backup button to save the configuration file on your PC.
Private Phonebook File	Select desired private phonebook file and press Update button to update. Press Backup button to save the phonebook file on your PC.
Public Phonebook File	Select desired public phonebook file and press Update button to update.

The upgrade process will start. It needs 2 minutes to finish, please do not turn off the power at this time. After updating successfully, the phone will reboot automatically.

TFTP/FTP Upgrade:

Server Type	Select the Server Type. (TFTP or FTP)
Server IP Address	IP address of TFTP or FTP Server.
Files Directory	Files directory where the software, configuration file, or phonebook file located.
Server User Name	Username of the account on the server.
User Password	Password of the account on the server.
Software File	Software file name on the server.
Configuration File	Configuration file name on the server.

Private Phonebook File	Private phonebook file name on the server.
Public Phonebook File	Public phonebook file name on the server.

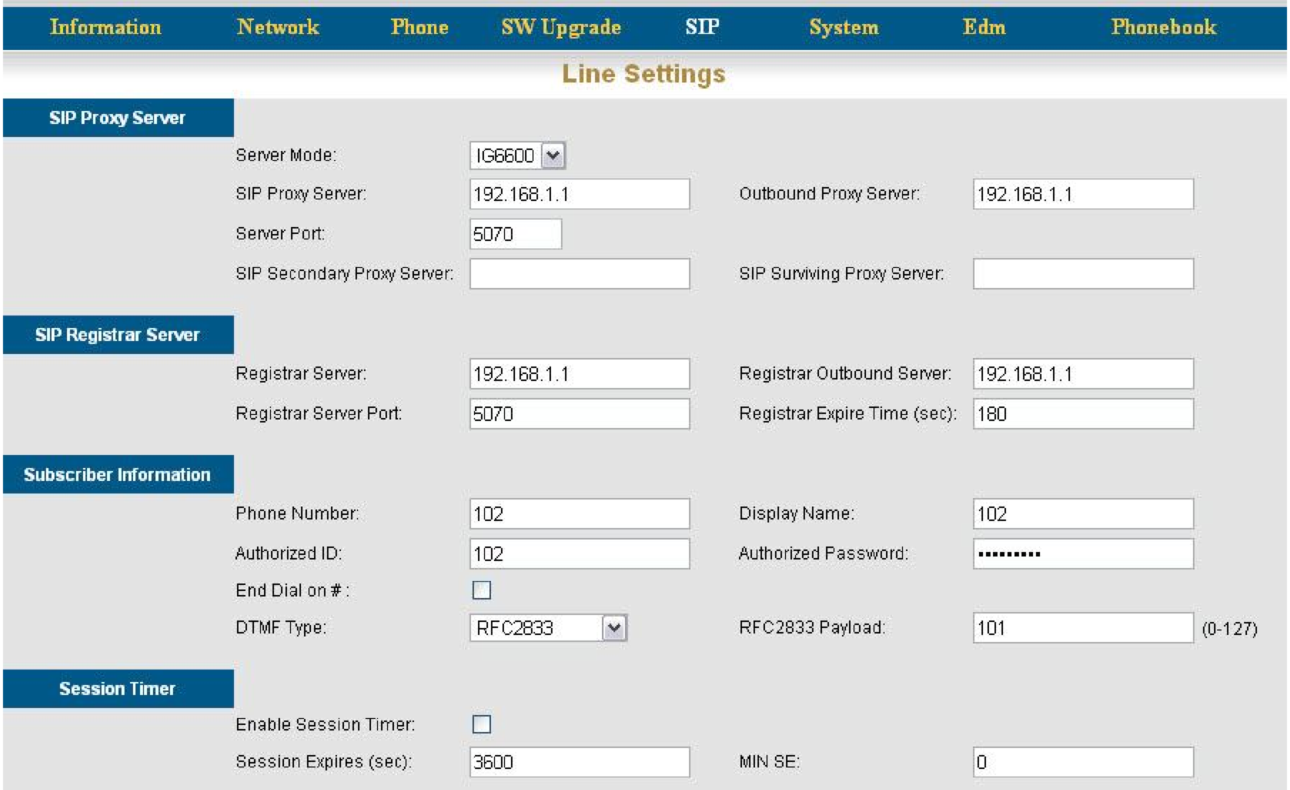
You can also use TFTP or FTP server to update the software, configuration file, or phonebook file. First, fill the correct file name which you desire to update. Second, press "Save Settings" button to save. Finally, press relative "Update" button, the update process will start.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.

Note: If there are errors while you updating the software and the phone can't work. Please press the "First Softkey" when phone just power on. Then you can see the hint on LCD. Set your TFTP server's IP to "192.168.1.11" and rename the correct software to "app.bin.gz". After pressing "ENTER" key, the phone will automatically update the software named app.bin.gz from fixed IP: 192.168.1.11.

SIP Page



SIP Proxy Server:

Field Name	*Function*
Server Mode	Different server mode could be set. There are two server mode supported: Normal and IG6600.
SIP Proxy Server	Set SIP proxy server.
Outbound Proxy Server	Set SIP outbound proxy server.
Server Port	Set SIP proxy service port.
SIP Secondary Proxy Server	Set SIP secondary proxy server.
SIP Surviving Proxy Server	Set SIP surviving proxy server.

SIP Registrar Server:

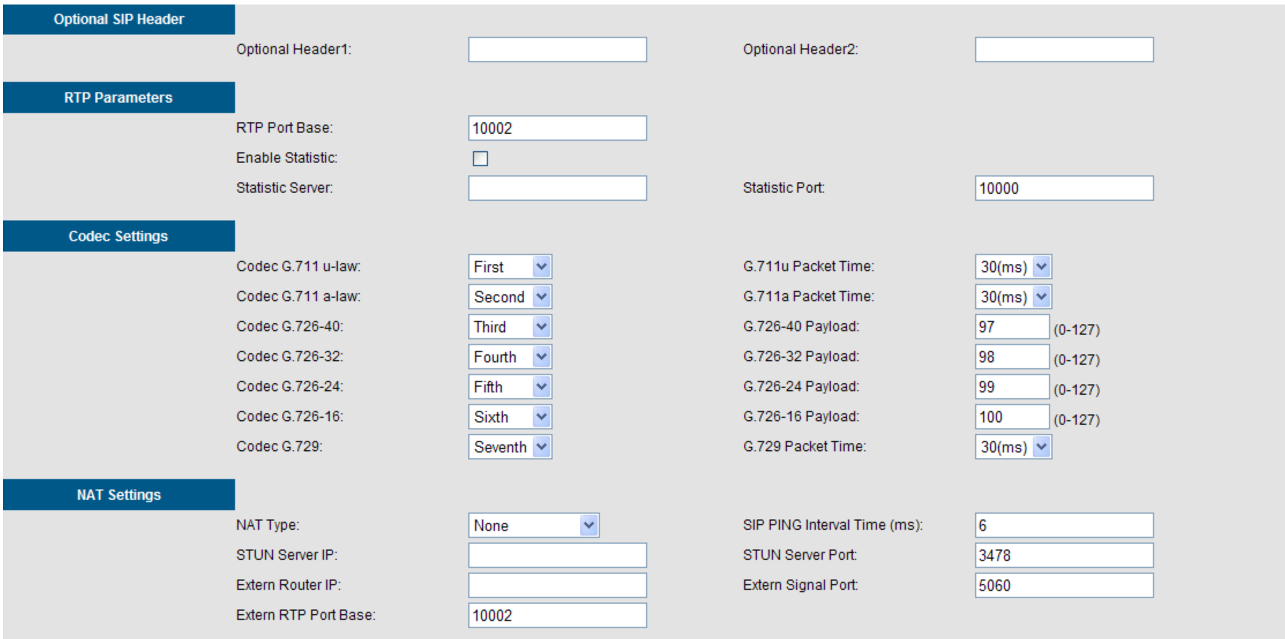
Registrar Server	Set SIP registrar server.
Registrar Outbound Server	Set SIP registrar outbound server.
Registrar Server Port	Set SIP registrar port.
Registrar Expire Time (sec)	Set SIP registrar expire time.

Subscriber Information:

Phone Number	Set the phone number.
Display Name	Set the display name.
Authorized ID	Set the authorized ID for SIP registration.
Authorized Password	Set the authorized password for SIP registration.
End Dial on #	Enable # key to end dialing.
DTMF Type	Select the way to send DTMF through in-band or out- band mode (RFC2833/SIP-INFO).
RFC2833 Payload	Set the RFC2833 payload number.

Session Timer:

Enable Session Timer	Turn on the session timer. When a session is established, phone will resend INVITE packets every half of this time for preventing from packet lost. The re-INVITES ensure that active sessions stay active and completed sessions are terminated.
Session Expires	Set expire time for session.
MIN SE	Set minimum session expiration time. It will notify server the minimum expiration time that you accepted in negotiation state. It is conveyed in the Min-SE header in the initial INVITE request.



Optional SIP Header:

Optional Header 1	Set value for SIP optional header 1.
Optional Header 2	Set value for SIP optional header 2.

RTP Parameters:

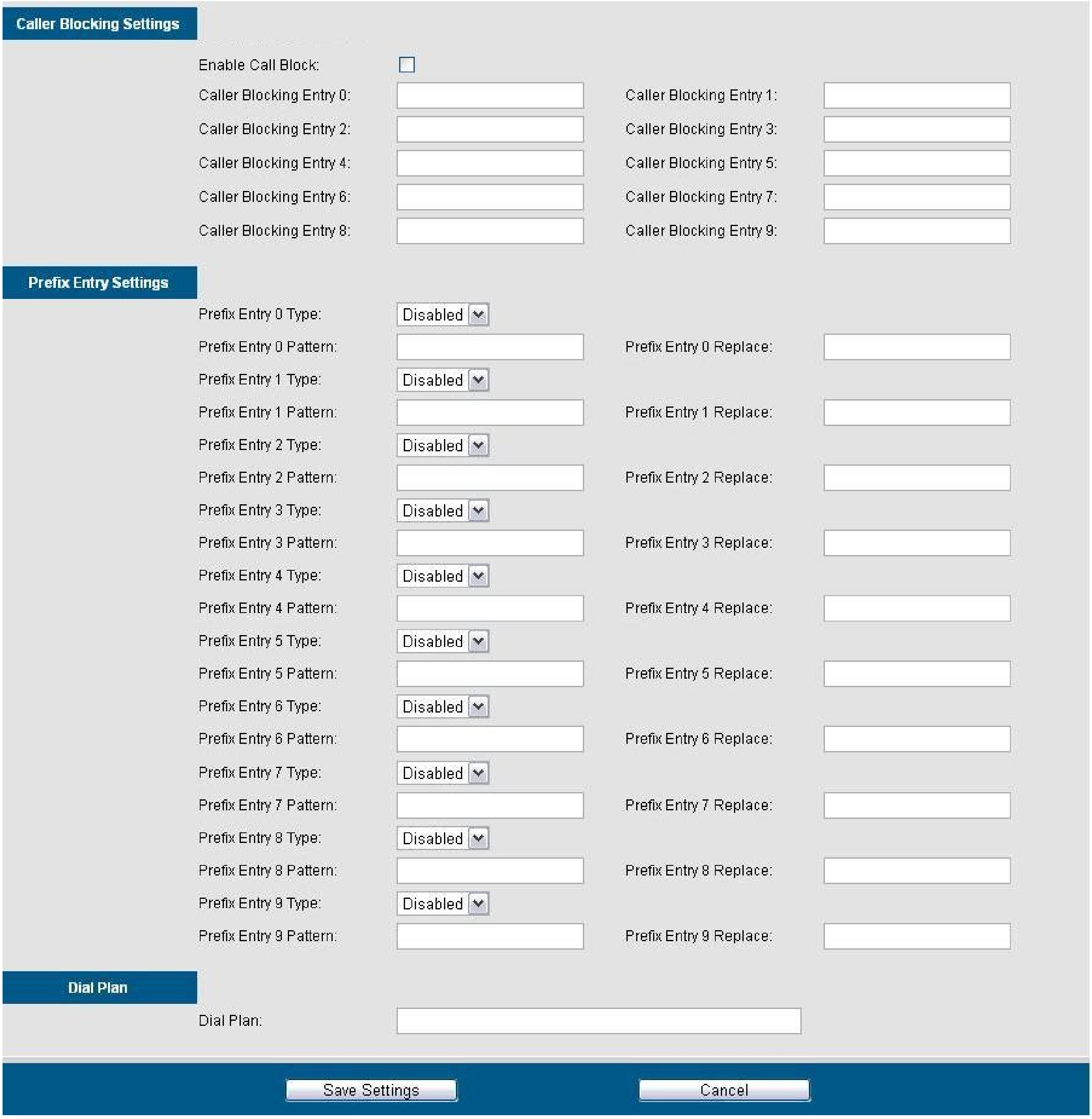
RTP Port Base	Set port base for RTP (Real-time Transport Protocol).
Enable Statistic	Enable statistic RTP server.
Statistic Server	Set statistic server.
Statistic Port	Set statistic port.

Codec Settings:

G.711u-law/a-law, G.726-40/32/24/16, G.729	Select the codec priority for RTP communications.
Packet Time	Select the relative packet time.

NAT Settings:

NAT Type	Select NAT type. There are 4 types for user to select, STUN, SIP PING, Port Mapping, and UDP Heartbeat.
SIP PING Interval Time (ms)	Set the SIP PING frequency.
STUN Server IP	Set STUN server IP address.
STUN Server Port	Set STUN server port.
Extern Router IP	Set external router address for port mapping.
Extern Signal Port	Set external SIP signaling port for port mapping.
Extern RTP Port Base	Set external RTP port base for port mapping.



Caller Blocking Settings:

Enable Call Block	Turn on the Call Block function.
Caller Blocking Entry 0~9	There are 10 blocking entries for user to set. When the caller ID of the incoming call matches a certain blocking entry, the call will be blocked. The remote party will hear busy tone.

Prefix Entry Settings:

Prefix Entry Type, Pattern, or Replace	If the "Prefix Entry Type" is "Replace" mode, the "Prefix Entry Pattern" will be replaced by the "Prefix Entry Replace". If the "Prefix Entry Type" is "Add" mode, the "Prefix Entry Replace" will be added behind the "Prefix Entry Pattern". For example, the "Prefix Entry Pattern" is 220 and the "Prefix Entry

	Replace" is 210. When you dial 220 with keyboard, the phone will dial 210 for "replace" mode or 220210 for "Add" mode.

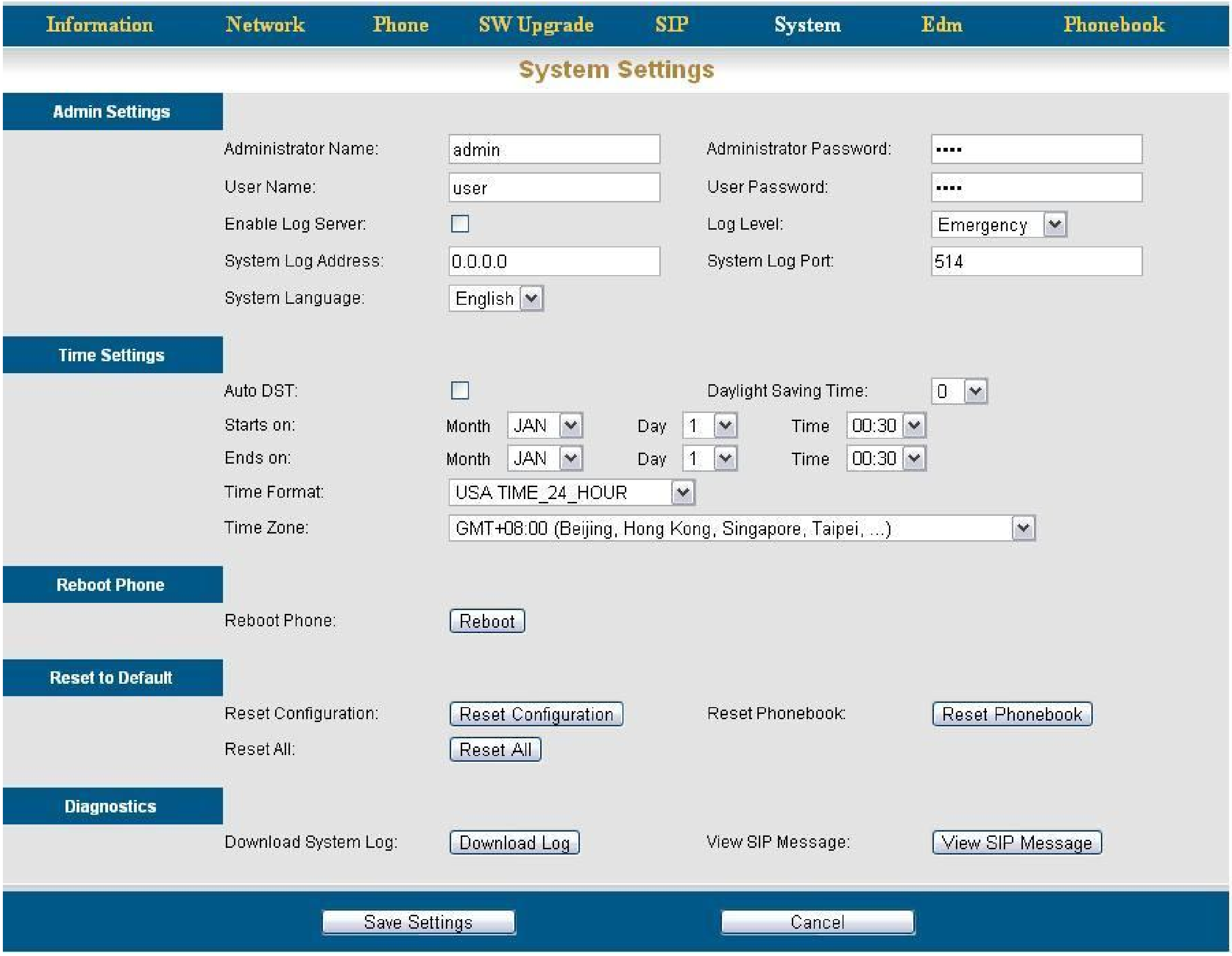
Dial Plan:

Dial Plan	Outgoing call which fit for the dial plan rule will be called out immediately, without waiting the dial timeout. It is convenient for users to make call quickly. There are several kinds of settings for dial plan: 1. "x" means any single digit from 0~9. 2. "[digit-digit]" means a valid range for a certain number position. 3. [digits] means acceptable numbers for a certain number position. 4. "|" is used for separate different dial plan items. For example, set dial plan as "60xx|30[6-9]|10[45]", if user dial 6000~6099, 306, 307, 308, 309, 104, and 105, IP phone will dial out it immediately. Otherwise, dial out the number after dial timeout.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.

System Page



Admin Settings:

Field Name	*Function*
Administrator Name	The administrator username to login phone's webpage.
Administrator Password	The administrator password to login phone's webpage.
User Name	The user username to login phone's webpage.
User Password	The user password to login phone's webpage.
Enable Log Server	Turn on the log server.
Log Level	Select log level.
System Log Address	Set IP address for system log server.
System Log Port	Set port for system log server.
System Language	Set system language.

The user's name is not allowed to the same as administrator's name. The administrator's and user's name should be set at least 1 character, not allowed to be null. The administrator's and user's password should be set at least 4 characters. When changing password in phone's configuration menu, less than 4 characters will be discard, not be saved.

The IP2061 provides System Log function. The administrator could assign IP address and port for system log server. There are 8 log levels for administrator to select.

Time Settings:

Auto DST	Turn on the auto DST (Daylight Saving Time) function.
Daylight Saving Time	Three time shifts could be set, -1, 0, and +1.
Starts on	Start time for DST.
Ends on	End time for DST.
Time Format	USA 24_HOUR, USA 12_HOUR, European 24_HOUR, and European 12_HOUR.
Time Zone	Set time zone.

Reboot Phone:

Reboot Phone	Reboot this IP Phone.

Reset to Default:

Reset Configuration	Reset Configuration to factory default.
Reset Phonebook	Clear all Phonebook data.
Reset All	Reset all data to default including Configuration, Call Logs, and Phonebook data.

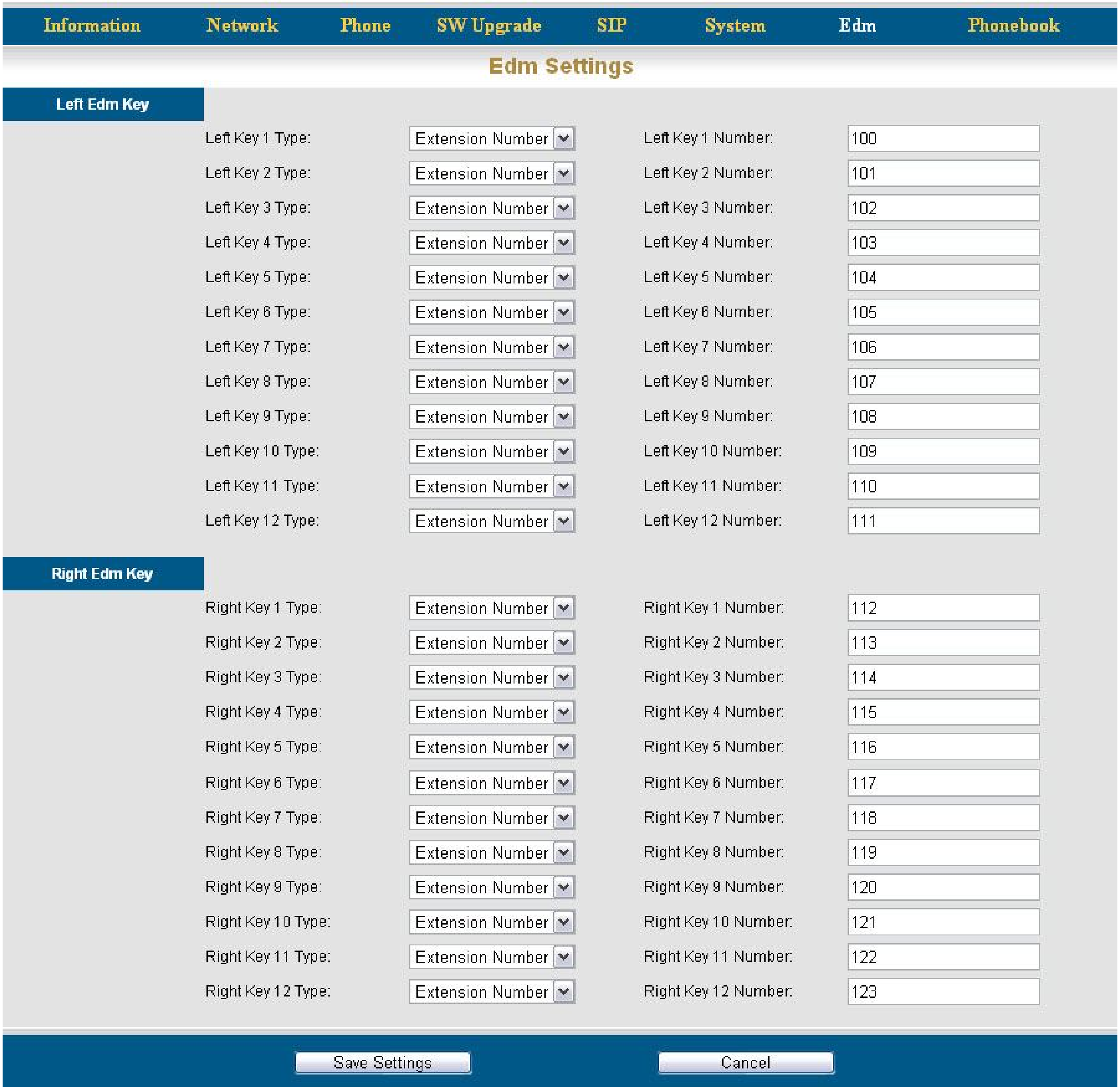
Diagnostics:

Download System Log	Press to download system log.
View SIP Message	Press to view SIP messages.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.

EDM Page



Each EDM (Extension Dial Module) has 24 keys.

Field Name	***Function***
Key Type	There are 9 types could be chosen: None, Extension Number, Trunk Number, Park number, Feature Code, Do Not Disturb, Live Record, Virtual Mailbox and Others. Phone will do the relative function when user pressing related line key.
Key Number	The string will be executed when the line key type is set to Extension Number, Trunk Number, Park number, Feature Code, Virtual Mailbox and Others.

The application for the EDM buttons is the same Line Keys.

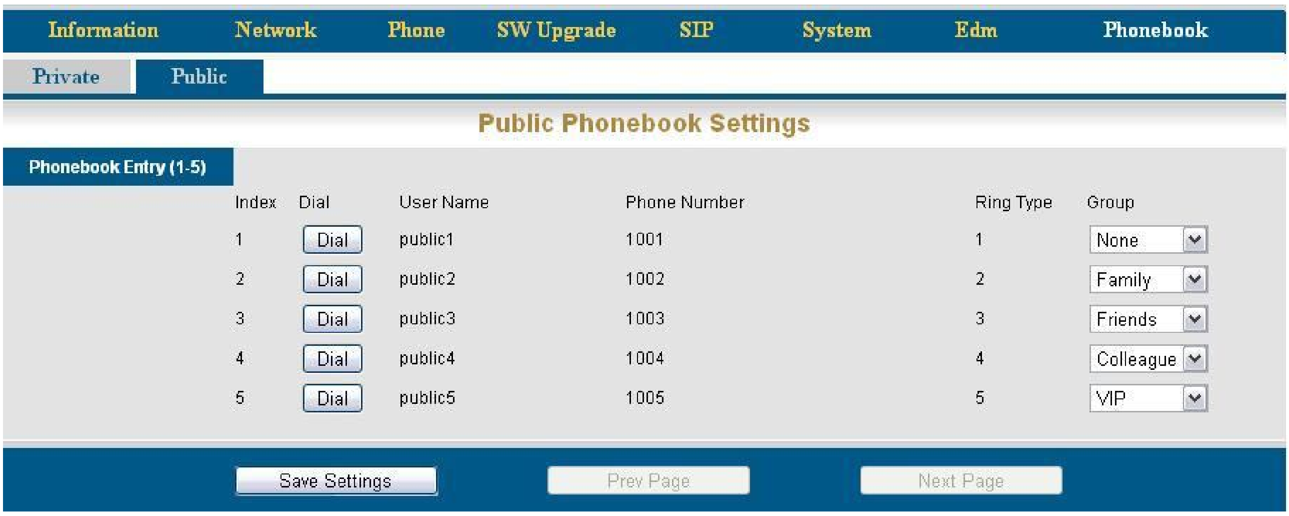
Phonebook

Private Phonebook:

This page lets you configure the private phonebook of your phone. You may:

- Edit up to 100 entries. Each page will just show 10 entries.
- Dial a phone number directly from the webpage.

Public Phonebook:



You can also check public phonebook when a public phonebook file is already uploaded. Up to 100 entries could be stored.

Phonebook Entry:

Field Name	***Function***
User Name	Name of your contact.
Phone Number	Phone number or SIP URI of your contact.

Ring Type	Select particular ring type for each entry.
Group	Select certain group for a phonebook entry.

To call your contact, simply click the corresponding Dial button and it will show the following dialog box (for example, if you click first entry to call out, the result will be the following):

Then press the "Dial" button in order to start dialing. Phone will dial out corresponding number automatically.

6. Features & Specifications

With highly integrated chip and sophisticated design, this IP2061 offers a variety of features. And it can provide high performance, reliable and quality voice communication for the users. The hardware specifications are list as follows for your reference.

Main Unit
-Dimension: 210 mm * 230 mm * 50 mm.
-Plastic material: ABS type.
-Support 30 ^{o}, 45 ^{o} degree stand angles.
-Can be mounted onto the wall.
-Support detachable handset and curled cord.

LAN and PC Ports
-Integrated 2 ports Ethernet switch.
-IEEE 802.3 10BaseT / 802.3u 100BaseTx compliant.
-Auto-negotiation with link speed and full/half duplex mode.
-Auto MDI/MDIX for both downlink and uplink auto-swapping.
-Support QoS IEEE 802.1p voice priority function.

Power Supply
-Input: 5VDC/2A power adapter.
-Support PoE (Power over Ethernet), IEEE 802.3af Device - Class 2 (7W).

Voice Handling
-Supports multiple Audio Codecs: G.711 a-law/ µ-law, G.726-40/32/24/16 (G.729 and G.723 can be supported if needed).
-Supports VAD (Voice Activity Detection) and CNG (Comfort Noise Generation).
- Volume adjustable for handset, headset, hands-free and ring.
-Support G.165 16ms line Echo Cancellation.
-Adaptive Jitter Buffering function supported.
-Hands-free talking supported.
-Support 4 calls.

Tone Function:
-DTMF tone generation.
-Side tone and good voice quality supported.
-Out-bound DTMF relay (RFC2833/SIP-INFO) support.
-Local tone support (Dial, Ring, Ring back, Busy and related tones).

Feature Access Codes:

The Feature Access Codes are applied between IP2061 and IG6600. It's used to activate/deactivate some user-specified functions.
These Feature Access Codes are accepted when the phone is at idle state.

Direct Call Forward

Forward all of the calls without regard to the extension status.

To Activate (Type: 0 – ICM; 1 – Outside; 2 – Both)

*21 + Type + Ext/VAA/ICD No

*21 + Type + * + (PSWD) + * + Outside Number

To Cancel

**21

Busy Call Forward

Forward the calls if the extension is busy.

To Activate (Type: 0 – ICM; 1 – Outside; 2 – Both)

*22 + Type + Ext/VAA/ICD No.

*22 + Type + * + (PSWD) + * + Outside Number

To Cancel

**22

No Answer Call Forward

Forward the calls if the extension doesn't answer the call after No Answer Time.

To Activate (Type: 0 – ICM; 1 – Outside; 2 – Both)

*23 + Type + Ext/VAA/ICD No + * + Time

*23 + Type + * + (PSWD) + * + Outside Number + * + Time

To Cancel

**23

DND Call Forward

Forward the calls if the extension enables DND.

To Activate (Type: 0 – ICM; 1 – Outside; 2 – Both)

*24 + Type + Ext/VAA/ICD No.

*24 + Type + * + (PSWD) + * + Outside Number

To Cancel

**24

Follow Me Call Forward

Forward calls at your extension to the extension where you are currently working.

To Activate (Type: 0 – ICM; 1 – Outside; 2 – Both)

*25 + Type + Ext No + * + Password

To Cancel

**25 + Ext No + * + Password

Call Fork

When extension gets an incoming call, the extension gets ringing. It rings another extension or rings an outside destination simultaneously.

(Type: 0 – ICM, 1 – Outside, 2 – Both; Destination: 1 – First; 2 – Second)

To Activate

*26 + Destination# + Type + Ext No
*26 + Destination# + Type + * + (PSWD) + * + Outside Number

To Cancel

**26 → cancel the both forking destination

**261 → cancel the first forking destination only

**262 → cancel the second forking destination only

DND (Do Not Disturb)

Extension users can enable DND to stop incoming calls from ringing at their phone.

To Activate

*4

To Cancel

**4

Call Pickup

Users can answer the calls at another extension. The feature allows you to easily access calls ringing via the feature access code.

*53 + Ext No.

COS Following

It changes the individual COS of the extension temporarily.

*55 + (phone number) + (phone's password)

Call Back Busy

When remote party is busy and hearing busy tone, press "6" to wait them call back. Pressing "*66" to delete the record.

To Activate

6

To Cancel

*66

Reset Feature Buttons

Reset all feature buttons to IG6600's setting.

*68 + (Password)

Reset To Default

Selected IG6600 extension features can be returned to default setting.

*69 + (Extension/Administrator password)

Feature Key Programming

To program the line keys as a PSTN, IP Trunk, Trunk Group number, Call-Park number or Extension number.

*70 + (Feature Key number: 01 – 28) + (Feature Key Type: 00 – 06) + Number

Feature Key Type:

00: Null; Number should be null.
01: Extension; Number can be an Extension number.
02: Trunk; Number can be a PSTN, IP Trunk or Trunk Group number.
03: Call-Park; Number can be a Park number.
04: Feature Key; Number can be a feature access code
05: Others; Number could be an outside phone number.
06. Do Not Disturb; Number should be null.
07: Live Record
08: Virtual Mailbox key; Number can be a Virtual number.

Service Mode Selection

Change Service Mode from Operator

*79 + (Service Mode, 0 – 3)

Service Mode:

0. Change the Service Mode
1: Day Mode
2: Night Mode
3: Time Mode)

Agent Log On/Off

It can control the status in an ICD group.

To Activate (Log On)

*91

To Cancel (Log Off)

**91

Phone Lock/Unlock

You can use the lock feature to prevent unauthorized "trunk calls" that being made from extension.

To Activate Phone Lock

*97 + (password)

To Cancel Phone Lock

**97 + (password)

Call Waiting

If it's enabled, it can allow other incoming calls when phone is in talk state. If it's disabled, other incoming caller will get busy tone when phone is in talk state.

To Activate

*98

To Cancel

**98

Page Allow/Deny

To allow/deny group paging or all paging.

To Activate Page Deny

*99

To Cancel Page Deny

**99

Hotline

To allow automatically access a given resource each time the extension goes off hook.

To Activate

9 + (any Number) + * + Time //Time: 0~8 seconds; 0: immediately

To Cancel

**9*

7. Troubleshooting

Symptom	*Check & Remedy*
No operation	● Check if the power adapter is properly connected. ● Check if the Ethernet cable is properly connected. ● If applicable, check if the PoE (Power over Ethernet) switch behind the IP phone is set to the right position.
No dial tone	● Check if the handset cord is properly connected. ● Check if the power adapter is properly connected.
Check network connection shown on LCD	● Check if the Ethernet cable is properly connected.
Cannot make call	● Check the status of your SIP registration or contact your supplier or ITSP for more information or assistance.
IP phone cannot receive any phone call	● Check if the Ethernet cable is properly connected ● Check the status of your SIP registration or contact your supplier or ITSP for more information or assistance.
Cannot connect to the configuration website of the IP phone	● Check if the Ethernet cable is properly connected. ● Check the IP address of the IP phone. ● Check if your firewall/NAT settings are correct.

8. Glossary

Acronyms

CODEC	Coder and Decoder of Voice
CNG	Comfort Noise Generation
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name Server
DTMF	Dual Tone Multiple Frequency
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
ISP	Internet Service Provider
ITSP	IP Telephony Service Provider
LAN	Local Area Network
MWI	Message Waiting Indication
PoE	Power over Ethernet (IEEE802.3af standard)
PPPoE	Point-to-point protocol over Ethernet
QoS	Quality of Service
NAT	Network Address Translation
RTP	Real Time Protocol
SIP	Session Initiation Protocol
SNTP	Simple Network Address Translation
TFTP	Trivial File Transfer Protocol

Terminology

10/100BASE-T	It's a LAN transmission line specification stipulated by IEEE. Transmission speed is 10 or 100 Mbps and the modulation technique is base-band modulation. The cable uses unshielded twisted pair, similar to a telephone wire. 10BaseT is an IEEE standard (802.3) for operating 10 Mbps Ethernet networks (LANs) with twisted pair cabling and a wiring hub.
Auto answer	In telephone call control: The capability of a machine to answer a ringing telephone without human intervention.
CODEC	The CODEC (CODER/DECODER) is a standard through which voice information can be encoded into data or decoded back to voice information. Both a Coder and Decoder are necessary on both sides of the telephone call since telephone calls occur simultaneously in both directions. Bandwidth is an extremely important factor in QOS (Quality of Service). MOS (Mean Opinion Score) is an attempt to make a quantifiable benchmark of voice quality. Below are examples of the CODEC, bit rate and mean opinion score: G.711 (toll quality) 64K MOS=4.1, G.726 16K (32K) MOS =3.8, G.729AB (cell phone quality) 8K MOS=3.7.
DHCP	Dynamic Host Configuration Protocol (DHCP): A utility that enables a server to dynamically assign IP addresses from a predefined list and limit their time of use so that they can be reassigned. Without DHCP, an IT Manager would have to manually enter in all the IP addresses of all the computers on the network. When DHCP is used, whenever a computer logs onto the network, it automatically gets an IP address assigned to it.
Diff-Serv	Differentiated Services: The Diff-Serv model divides traffic into a small number of classes to provide quality of service (QoS). One of QoS in internet.
DNS	Domain Name Service (DNS): A server/program that translates URLs to IP addresses by accessing a database maintained on a collection of Internet servers. The program works behind the scenes to facilitate surfing the Web with alpha versus numeric addresses. A DNS server converts a name like mywebsite.com to a series of numbers like 107.22.55.26. Every website has its own specific IP address on the Internet. Typically one or more DNS servers is located in an IP network.
DTMF	Dual Tone Multi-Frequency (DTMF): The type of audio signals generated when you press the buttons on a touch-tone telephone. Can be used in inbound (after voice channel connected) and outbound (before voice channel connected) application.
Ethernet	International standard networking technology for wired implementations. Basic 10BaseT networks offer a bandwidth of about 10 Mbps. Fast Ethernet (100 Mbps) and Gigabit Ethernet (1000 Mbps) are becoming popular.
G.711	64 kbps PCM half-duplex codec (high quality, high bandwidth, minimum processor load)
IP	Internet Protocol (IP) is located at 3rd layer of ISO network model. A set of rules used to send and receive messages at the Internet address level. IP protocol is widely used in Internet and LAN networks. The purpose is to deliver data between computing equipment over the network. The protocol is generally effective but does not guarantee complete and accurate data communications.

IP address	A 32-bit number that identifies each sender or receiver of information that is sent across the Internet. An IP address has two parts: an identifier of a particular network on the Internet and an identifier of the particular device (which can be a server or a workstation) within that network. A number used to identify the location of a host device. It is expressed in numeric dot notation (e.g. 202.203.27.31).
MAC address	Media Access Control address (MAC address): It is a unique identifier assigned to most network adapters or network interface cards by the manufacturer for identification, and used in the Media Access Control protocol sub-layer. It may also be known as an Ethernet Hardware Address or physical address. In TCP/IP networks, the MAC address of a subnet interface can be queried with the IP address using the Address Resolution Protocol (ARP) for Internet Protocol. On broadcast networks, such as Ethernet, the MAC address uniquely identifies each node and allows frames to be marked for specific hosts. It thus forms the basis of most of the Link layer (OSI Layer 2) networking upon which upper layer protocols rely to produce complex, functioning networks.
NAT	Network Address Translation (NAT): A network capability that enables a houseful of computers to dynamically share a single incoming IP address from a dial-up, cable or xDSL connection. NAT takes the single incoming IP address and creates new IP address for each client computer on the network.
Proxy server	Used in larger companies and organizations to improve network operations and security, a proxy server is able to prevent direct communication between two or more networks. The proxy server forwards allowable data requests to remote servers and/or responds to data requests directly from stored remote server data.
PSTN	Public Switched Telephone Network (PSTN): The worldwide voice telephone network.
RJ-45	Standard connectors used in Ethernet networks. Even though they look very similar to standard RJ-11 telephone connectors, RJ-45 connectors can have up to eight wires, whereas telephone connectors have only four.
RTP/RTCP	Real-Time Protocol/Real-Time Control Protocol (RTP/RTCP): IETF specifications for audio and video signal management. Allows applications to synchronize and spool audio and video information. RTP is specifically concerned with the dependable transmission of latency-sensitive traffic across the network and is involved in using time stamping to determine network jitter tolerance and makes sure that voice packets are arriving in order.
SIP	Session Initiation Protocol (SIP): A protocol that provides telephony services similar to H.323, but is less complex and uses fewer resources. SIP is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users.
Subnetwork or Subnet	Found in larger networks, these smaller networks are used to simplify addressing between numerous computers. Subnets connect to the central network through a router, hub or gateway. Each individual wireless LAN will probably use the same subnet for all the local computers it talks to.
TCP/IP	Internet Standard Protocol: The underlying technology behind the Internet and communications between computers in a network. The first part, TCP, is the transport part, which matches the size of the messages on either end and guarantees that the correct message has been received. The IP part is the user's computer address on a network. Every computer in a TCP/IP network has its own IP address that is either dynamically assigned at startup or permanently assigned.

	All TCP/IP messages contain the address of the destination network as well as the address of the destination station. This enables TCP/IP messages to be transmitted to multiple networks (subnets) within an organization or worldwide.
VAD	Voice Activity Detection (VAD) helps save bandwidth during calls. Examples: When you making a VoIP call and your not speaking and your listening, that silence is still taking up bandwidth during the call. When silence is detected by VAD software over a predetermined length of time, it sends silent packets that inform other VAD enabled systems to stop holding the bandwidth for these empty packets.
VoIP	Voice Over Internet Protocol: VoIP is based on the principal of transmitting digitized voice packets over networks. Basically, VoIP consists of converting voice signals into streams of digital packets and sending those packets of data through an IP-constructed network environment. VoIP can work in both LAN (local area network) and WAN (wide area network) environments for intranetwork or internetwork communication between VoIP channel users. Routers and switches and other special compression protocols direct the packetized voice data to their destination IP address. VoIP can be less expensive than voice transmission using standard analog packets over POTS (Plain Old Telephone Service). It allows telephone calls, faxes, or overhead paging to be transported over an existing IP data network topology.